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SUMMARY TECHNICAL REPORT

OF THE

NATIONAL DEFENSE RESEARCH COMMITTEE

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SUMMARY TECHNICAL REPORT OF DIVISION 17, NDRC

VOLUME 3

TRANSMISSION AND RECEPTION OF SOUNDS UNDER COMBAT CONDITIONS

OFFICE OF SCIENTIFIC RESEARCH AND DEVELOPMENT
VANNEVAR BUSH, DIRECTOR

NATIONAL DEFENSE RESEARCH COMMITTEE
JAMES B. CONANT, CHAIRMAN

DIVISION 17
GEORGE R. HARRISON, CHIEF

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WASHINGTON, D. C., 1946

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NOTES ON THE ORGANIZATION OF NDRC

The duties of the National Defense Research Committee were (1) to recommend to the Director of OSRD suitable projects and research programs on the instrumentalities of warfare, together with contract facilities for carrying out these projects and programs, and (2) to administer the technical and scientific work of the contracts. More specifically, NDRC functioned by initiating research projects on requests from the Army or the Navy, or on requests from an allied government transmitted through the Liaison Office of OSRD, or on its own considered initiative as a result of the experience of its members. Proposals prepared by the Division, Panel, or Committee for research contracts for performance of the work involved in such projects were first reviewed by NDRC, and if approved, recommended to the Director of OSRD. Upon approval of a proposal by the Director, a contract permitting maximum flexibility of scientific effort was arranged. The business aspects of the contract, including such matters as materials, clearances, vouchers, patents, priorities, legal matters, and administration of patent matters were handled by the Executive Secretary of OSRD.

Originally NDRC administered its work through five divisions, each headed by one of the NDRC members. These were:

Division A—Armor and Ordnance
Division B—Bombs, Fuels, Gases, & Chemical Problems
Division C—Communication and Transportation
Division D—Detection, Controls, and Instruments
Division E—Patents and Inventions

In a reorganization in the fall of 1942, twenty-three administrative divisions, panels, or committees were created, each with a chief selected on the basis of his outstanding work in the particular field. The NDRC members then became a reviewing and advisory group to the Director of OSRD. The final organization was as follows:

Division 1—Ballistic Research
Division 2—Effects of Impact and Explosion
Division 3—Rocket Ordnance
Division 4—Ordnance Accessories
Division 5—New Missiles
Division 6—Sub-Surface Warfare
Division 7—Fire Control
Division 8—Explosives
Division 9—Chemistry
Division 10—Absorbents and Aerosols
Division 11—Chemical Engineering
Division 12—Transportation
Division 13—Electrical Communication
Division 14—Radar
Division 15—Radio Coordination
Division 16—Optics and Camouflage
Division 17—Physics
Division 18—War Metallurgy
Division 19—Miscellaneous
Applied Mathematics Panel
Applied Psychology Panel
Committee on Propagation
Tropical Deterioration Administrative Committee

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NDRC FOREWORD

As EVENTS of the years preceding 1940 revealed more and more clearly the seriousness of the world situation, many scientists in this country came to realize the need of organizing scientific research for service in a national emergency. Recommendations which they made to the White House were given careful and sympathetic attention, and as a result the National Defense Research Committee [NDRC] was formed by Executive Order of the President in the summer of 1940. The members of NDRC, appointed by the President, were instructed to supplement the work of the Army and the Navy in the development of the instrumentalities of war. A year later, upon the establishment of the Office of Scientific Research and Development [OSRD], NDRC became one of its units.

The Summary Technical Report of NDRC is a conscientious effort on the part of NDRC to summarize and evaluate its work and to present it in a useful and permanent form. It comprises some seventy volumes broken into groups corresponding to the NDRC Divisions, Panels, and Committees.

The Summary Technical Report of each Division, Panel, or Committee is an integral survey of the work of that group. The first volume of each group's report contains a summary of the report, stating the problems presented and the philosophy of attacking them, and summarizing the results of the research, development, and training activities undertaken. Some volumes may be "state of the art" treatises covering subjects to which various research groups have contributed information. Others may contain descriptions of devices developed in the laboratories. A master index of all these divisional, panel, and committee reports which together constitute the Summary Technical Report of NDRC is contained in a separate volume, which also includes the index of a microfilm record of pertinent technical laboratory reports and reference material.

Some of the NDRC-sponsored researches which had been declassified by the end of 1945 were of sufficient popular interest that it was found desirable to report them in the form of monographs, such as the series on radar by Division 14 and the monograph on sampling inspection by the Applied Mathematics Panel. Since the material treated in them is not duplicated in the Summary Technical Report of NDRC, the monographs are an important part of the story of these aspects of NDRC research.

In contrast to the information on radar, which is of widespread interest and much of which is released to the public, the research on subsurface warfare is largely classified and is of general interest to a more restricted group. As a consequence, the report of Division 6 is found almost entirely in its Summary Technical Report, which runs to over twenty volumes. The extent of the work of a division cannot therefore be judged solely by the number of volumes devoted to it in the Summary Technical Report of NDRC: account must be taken of the monographs and available reports published elsewhere.

The research work of Division 17 included a wide variety of projects, ranging from the detection of land mines to the characteristics of the human ear, from helium purity indicators to the telemetering of strain gauges, from odographs to sound-ranging devices. It is a tribute to the broad knowledge of the Division Chiefs—Paul Klopsteg and, later, George R. Harrison—and to the versatility of the men who worked under them that so diverse a program was handled so competently.

A considerable portion of the work of Division 17 had to do with the shattering noise of modern war, and answers were sought and supplied to such questions as: How much noise can a human being stand? What clues must the human ear have in order to understand a spoken message? How much distortion can be tolerated? These and other phases of the Division's work are dealt with in the Summary Technical Report prepared under the direction of the Division Chief and authorized by him for publication.

The diversity of the Division's projects made it inevitable that its staff should be composed of men with many types of scientific training and that the Division should draw on contractors with a wide range of experience and skills. The studies of noise, in particular, meant that the technical staff must include physicists, acousticians, and psychologists. For the ability and devotion of these men of many aptitudes we express our gratitude.

VANNEVAR BUSH, Director
Office of Scientific Research and Development

J. B. CONANT, Chairman
National Defense Research Committee

FOREWORD

THE APPLIED PHYSICS DIVISION of the Office of Scientific Research and Development [OSRD] was organized late in 1942 under the chairmanship of Dr. Paul E. Klopsteg, who was responsible for the work of the Division until shortly before the completion of its work when other duties required his full attention. Most of the projects which had been initiated by the Instruments Section of the National Defense Research Committee [NDRC] during 1940 and 1941 and which were not concerned with optics were turned over to the Applied Physics Division on its inauguration. Dr. Klopsteg as Chief and Dr. E. A. Eckhardt as Deputy Chief went with them from the Instruments Section to the new Division.

The Summary Technical Report which is presented in these volumes thus covers the accomplishments of projects set up by both Section D-3 and Division 17. The work of the Division covered a very wide range of fields. The term *Applied Physics* served in lieu of a more descriptive name for a Division which was in fact the one to which was assigned any scientific problem which did not properly come under one of the other Divisions of NDRC.

Actually the Division was an association of three Sections having rather dissimilar responsibilities and fields of activity. In setting up these Sections it was necessary to group the projects already under way into a small number of coherent categories, and those chosen were Sound, Electricity, and General Instrumentation. The work of the Division consisted entirely of the integrated efforts of these three Sections, whose membership will be found listed on a succeeding page.

For more detailed reports on the technical work of the Division than are contained herewith the detailed contractors' reports of Division 17 should be consulted, and appropriate reference to these have been made throughout the present volumes. The results obtained are also presented in less technical form in that volume of the history of OSRD entitled *Optics and Applied Physics in World War II.*

GEORGE R. HARRISON
Chief, Division 17

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PREFACE

The research and development program of Division 17, NDRC, was concerned with those problems in physics not specifically covered in other Divisions of NDRC. As the result, the Division fell heir to a myriad of miscellaneous problems of a physical nature which, in themselves, were not often interrelated. It would have been exceedingly difficult, if not impossible, for Division 17 to set up within itself a sufficient number of Sections to deal specifically with all the various classes of problems which fell under their jurisdiction. Therefore, the projects of the Division were assigned to one of three Sections —Section 17.1, Instruments; Section 17.2, Electrical Equipment; and Section 17.3, Acoustics— whose broad titles permitted a general, even if somewhat loose classification. It was not always easy to decide under which of these three broad categories a given project should be placed. In these cases, considerations such as immediate convenience and availability of experienced personnel were often the determining factors.

The Summary Technical Report describing the activities of Division 17 is presented in four volumes. In an attempt to achieve a little greater uniformity of subject matter, the projects were organized within the various volumes without regard to their Section classification. Consequently, there is, on the whole, little relationship between volume and Section number. Because of the varied problems dealt with in the Division's program, very little continuity is to be found from chapter to chapter in any volume. Each chapter attempts to summarize independently the results of a particular project.

Since there were a large number of diversified projects in Division 17, it was impossible to do justice to each, even in summary. It is not intended that the importance of any project described herein should be judged by the amount of page space allotted to it. Naturally, certain problems involved more research and development than others before they could be brought to a successful conclusion. In many cases, this is reflected in the Summary Technical Report. On the other hand, the presentation of the projects may mirror the amount of enthusiasm of the individual author at the time of writing. Therefore, the reader who desires more than a broad panorama of the Division's activities is referred to the Microfilm Index for more complete details.

Volume 3 of the Division 17 Summary Technical Report deals with the *Transmission and Reception of Sounds under Combat Conditions.* This entire program was carried out at the Psycho-Acoustic and Electro-Acoustic Laboratories of Harvard University. As a result, the research on this program was more closely integrated and coordinated than many of the other Division 17 programs which were often divided among a number of different contractors. It was only natural, then, that the report summarizing the results of the investigations of the two Harvard Laboratories should be a single volume. This is especially true since the chapters bear a logical relation to each other and present a fairly complete picture of the problems involved in the transmission and reception of sounds under conditions of combat. There were many other projects in Division 17 on the transmission, reception, and simulation of sound, but they have been reported in Part II of Volume 2 rather than here. These other projects were mainly concerned with the simulation of combat noises and sonic deception; hence, the problems of transmission and reception, while important, were more of a secondary nature than those dealt with in this volume, where they are of fundamental importance.

All the chapters of this volume, except the Introduction, were written by G. A. Miller and F. M. Wiener of the Harvard Psycho-Acoustic Laboratory under the guidance of S. S. Stevens, its Director. These men merit high praise for their conscientious efforts in preparing their manuscript, and to them the Editor of this volume wishes to express his deep appreciation.

CHAS. E. WARING
Editor

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Chapter 1

INTRODUCTION

1.1 THE SCIENCE OF SOUND IN WARTIME

SOUND IS an important part of man's wartime environment. Sound waves in air, water, and other media are important and sometimes useful as physical phenomena, but mostly they are significant because men have ears to hear them. On waves of sound men "pass the word," a process which forms the backbone of effective military coordination.

Troops, tanks, ships, planes are welded into effective working units by means of speech, conveyed either directly through the air or over radios, interphones, or telephones. But modern warfare is noisy. To the cracking of rifle fire and the blast of artillery has been added the more sustained din of innumerable mechanized vehicles and devices. The problems of reducing noise, protecting personnel, avoiding detection by the enemy or confusing his intelligence are issues that must be faced by a modern armed force.

Back in 1940, when national defense was becoming the key to the national consciousness, there was mounting evidence that the science and art of military acoustics was in many particulars inadequate to the threatened demands. It was becoming increasingly clear that much of our communication equipment had too little power, too much distortion, and too great a vulnerability to noise. Tanks and airplanes were becoming noisier and soldiers were being threatened with deafness by din and blast. Special acoustic problems and applications were coming to the fore: direction finding by means of sound waves, acoustic warning and signaling systems, sonic airspeed indicators, etc.

To this state of affairs, the answer, of course, was research—scientific inquiry into the conditions of use, the causes of failure, the specifications for success. Take, for example, the problem of noise. Guns and motors presented hazards to voice communication seldom considered in peacetime laboratories. What does the terrific noise of modern warfare do to men? What levels of noise can be tolerated? How can excessive noise be reduced and controlled? How loud does speech have to be for men to hear it above the racket? When the problem of noise was taken into the laboratory, scientists determined the characteristics of typical noises, simulated them, and studied their effects. Often this required the development of new techniques or the revision of old ones. Physicists and psychologists worked side by side for five years on these problems, and the job was still not complete when the war ended.

It was quickly recognized that the human factor in communication is crucial. What clues must the human ear have to hear a spoken message, and what is the nature of the human voice itself? How much distortion can be tolerated by the listener? How loud must the speech be to stand out above the noise? And even when the voice and ear were accounted for, there was still the very human demand for comfort and convenience. Earphone sockets must exclude noise, but they must not crush the listener's ears. Microphones must be held against the lips, but they should not occupy the talker's hands. The human factor again appeared when it was necessary to select and train communication personnel. And even when the war was over the human problems of communication were not at an end, for deafened veterans needed hearing aids and general rehabilitation. At every point the experimental psychologist was called upon to effect an harmonious union of man and machine.

The recognition of the human factor would have been of little value, however, if trained physicists and acousticians had not been available to implement the psychologist's recommendations with adequate equipment. This work did not end with the evaluation of existing or proposed equipment, but on many occasions required the engineering of a new device. Even when the device had been accepted, it was often necessary to assist manufacturers in setting up testing methods and production short cuts. In addition to development, more difficult

problems of measurement had to be solved. When airplanes began reaching altitudes of 40,000 ft, communication failed. Reduced pressure lowered the sensitivity of microphones and earphones, changed the nature of the spoken word. It was necessary to devise methods of measuring performance at high altitudes and at the same time to make practical suggestions for improvements.

These are some of the problems discussed in detail in the following 18 chapters. These chapters have been organized around specific problems, rather than around specific contracts or chronological sequence. It is hoped that these chapters will form an introduction to the problem of reliable military voice communication and to the present state of our knowledge about it. Fundamental information is occasionally introduced in order to make the story more complete, and the physical and psychological aspects are treated together in summary as they were studied together in war. Since the organization and administration of the research is largely ignored in the following chapters, these aspects are briefly summarized in the following section.

1.2 THE THREE LABORATORIES

Unlike most Office of Scientific Research and Development [OSRD] projects, which fitted relatively neatly into one or another of the standard scientific fields, the research program in communications and airborne acoustics at Harvard was peculiarly an interdisciplinary venture. Problems related to sound and communications arise in almost every wing of the military structure: in airplanes, on shipboard, in tanks, in submarines, in diving suits, in pillboxes, and in the rehabilitation clinics that follow as an aftermath of war. Furthermore, since both gadgets and men are involved in these problems, their solution is not a task that can be entrusted to any one group nor to any one science. It is a matter for physicists and engineers on the one hand and for psychologists and physiologists on the other. And some of the problems call for the services of still other specialists, such as phoneticians and students of speech.

There were employed about 40 research men trained primarily in acoustics, electronics, and related subjects and about 35 scientists whose specialty was psychology, physiology, or speech.[a] From the fall of 1940 almost until the end of the war the work was divided principally between two laboratories, the Electro-Acoustic Laboratory, under the direction of L. L. Beranek, and the Psycho-Acoustic Laboratory, under the direction of S. S. Stevens. The philosophy of operation called for a certain division of labor, according to which the purely physical and electronic problems were pursued by the group at the Electro-Acoustic Laboratory, while those problems arising from the fact that a human being is part of the total circuit were made the specialty of the Psycho-Acoustic Laboratory. Actually, and fortunately, this division was never sharply maintained, and before the final bomb fell some of the physicists were deep in psychophysics and some of the psychologists were engineering new technical devices.

Early in 1945 a third laboratory was established to work out better methods of "passing the word" on shipboard, especially in the Combat Information Center, the nerve center of a ship. The Systems Research Laboratory operated mainly at a special field station set up at Beavertail Point, Jamestown, Rhode Island, where radar and communication gear was assembled to duplicate shipboard installations. The research staff included about 20 research men, mainly psychologists and physicists, but also a sprinkling of specialists in the field of "time and motion" engineering who were charged with the diagnosis and remedy of inefficiency in the activities and movements of the operating personnel at battle stations.

Partly because of the tendency of these projects to straddle the conventional scientific boundaries, and partly because they got off to an early start more than a year before Pearl Harbor, they had a varied and interesting contractual history with the United States Government and its agencies. It was never entirely clear to which division of OSRD the Harvard group ought most logically to be attached. It

[a] Under OSRD Contract OEMsr-658.

ended up with Division 17 (Physics) Section 3 (Acoustics), whose chief was Harvey Fletcher. On the other hand it undertook joint enterprises with Division 13 (Communications) and with the Applied Psychology Panel, and it might reasonably have had its primary association with either of these administrative units.

The Harvard projects got their start when the Army was seeking assistance with the problem of noise in airplanes. The question was not only how to reduce the noise in military aircraft but how to gauge its importance as a factor affecting the efficiency of pilots and aircrews. A Committee on Sound Control of the National Research Council [NRC] was organized, under the chairmanship of P. M. Morse, and this committee supervised the early project set up at Harvard. Funds were supplied by a contract[b] between the National Defense Research Committee [NDRC] of the Council of National Defense and the National Academy of Sciences on behalf of its agency, the National Research Council. A second contract[c] carried the project through the succeeding year and provided for an enlarged staff and a new program directed at problems of interphone communication. At the end of this period a new contract[d] was written directly between OSRD and Harvard University, and the NRC Committee on Sound Control ceased functioning. The supervision of this contract fell to NDRC Division C (Communications), Section 9 (P. M. Morse, Chairman), where it remained until the final reorganization of NDRC in January 1942. Under this reorganization the project landed in its final resting place, NDRC Section 17.3.

The last-mentioned contract[d] and its supplements remained in force until the end of January 1946. It provided a continually expanding series of performance clauses, based on a total of 14 Army and Navy directives, and resulted in the body of research outlined in the succeeding chapters.

[b] Contract NDCrc-52, effective November 10, 1940, to June 30, 1941.

[c] OEMsr-39.

[d] OEMsr-658.

The final windup caught many of the more basic research projects in midstream. Their abrupt termination would have meant the forfeit of a substantial return on a large investment of public funds, for answers almost attained would have been more or less permanently lost as the research personnel scattered to other jobs. It was at this point that the newly created Office of Naval Research of the U. S. Navy demonstrated its wisdom and foresight. Organized to sponsor fundamental research of interest to the national defense, this office set about to preserve those parts of the OSRD activities which could be turned toward basic long-range investigations of permanent significance. The entire research program under OEMsr-658 obviously could not be continued intact: it placed too large an emphasis on immediate, practical, wartime problems and it was staffed by men who had perforce to return to their peacetime occupations. The nucleus of basic problems stemming from the more practical wartime issues provided, however, the subject for a contract between the U. S. Navy and Harvard University.[e] The part of the program concerned with acoustics and electronics is continuing under Project Order I of this new contract, and Project Order II provides for a program of psycho-acoustic studies, with special emphasis on basic research in hearing and communications. Under this contract the Psycho-Acoustic Laboratory is bringing to completion the more significant researches which it initiated under OSRD, and it is embarking on fundamental inquiries not feasible under the stress of war.

Under another Navy contract, with Johns Hopkins University, the program initiated by the Systems Research Laboratory is also being continued and extended to cover basic research on information systems. The Hopkins program is under the direction of C. T. Morgan, who served during the war as Technical Aide to NDRC Section 17.3.

[e] Contract N5ori-76.

Chapter 2

SOUND CONTROL

2.1 MEASUREMENT OF AMBIENT NOISE

IN ORDER to assess quantitatively the effect of ambient noise on military personnel and to devise methods of reducing acoustic stress, it is necessary to make physical measurements of the noise intensities and spectra existing in representative military environments. In the early phases of the research carried out at the Electro-Acoustic and Psycho-Acoustic Laboratories in 1941 and 1942 it became evident that noise of battle has its most serious effect on voice communications. Combat vehicles, such as aircraft, ships, and tanks, are typical cases where the detrimental effect of ambient noise on voice communications is striking. The study of this problem formed a sizable part of the total effort of the two laboratories until the close of World War II. Several chapters are devoted to it.

The effect of noise on the psychomotor efficiency of personnel in battle was studied at the Psycho-Acoustic Laboratory and forms the subject matter of Section 2.5. Temporary hearing loss in military personnel after exposure to noise is quite common unless adequate precautions are taken (see Section 2.6).

The problems of meaningful measurement of levels and spectra of ambient noise and methods for their prediction and reduction have been studied extensively at the Electro-Acoustic Laboratory since 1941 and form the subject matter of the first four sections of this chapter.

The noise in combat vehicles of various kinds shows certain typical characteristics, since the physical processes of its generation are of a similar character. It can be analyzed into the following general components.

1. Noise produced by rotating or reciprocating mechanisms of propulsion. The noise spectrum consists of discrete components and their harmonics. The fundamental frequencies are of the order of 100 c. The harmonics decrease in magnitude with increasing order.

2. Noise produced by impact (tank treads, gear transmissions) or noise produced by air turbulence (aircraft) and similar causes. The spectrum of this noise approaches the continuous type and is, in most cases, the chief contributor from medium frequencies upward. In general, the spectrum level[2] decreases with frequency.

This section deals chiefly with the intensities and spectra of noise in typical combat vehicles and the methods and apparatus used to measure them. *Consideration of battle noise of other types, such as gunfire, explosions, etc., has been deliberately omitted.* Because of the transient nature and violence of these phenomena, their measurement, their reproduction in the laboratory, and their quantitative evaluation are difficult. Available field data are scant.

2.1.1 Methods and Apparatus

The apparatus suitable for noise analyses in combat vehicles and elsewhere consists essentially of a *microphone*, a suitable variable *band-pass filter*, an *amplifier*, and an *indicating instrument*.

The microphone is of the wide-range dynamic type[a] which has been calibrated in terms of the free-field sound pressure in a random sound field. This is the sound pressure which would be measured by a "point" microphone immersed in a random (diffuse) sound field. This type of sound field corresponds fairly closely to the sound fields commonly encountered.

The random incidence calibration is arrived at approximately by computational means, using free-field calibrations for progressive sound waves of various angles of incidence secured in an anechoic chamber of sufficient quality. All measured sound-pressure levels are expressed in decibels referred to the ASA reference level[10] of 0.0002 dyne cm^{-2}.

The variable band-pass filter covers the audio-frequency range from about 30 to 10,000 c. To analyze the various discrete low-frequency con-

[a] Western Electric instruments Types 630-A and 633-A have been used successfully.

tributions to the noise spectrum, a continuously variable filter arrangement of the wave-analyzer type can be used. Such an analysis, while time-consuming, permits interpretation of the various prominent spectral components in terms of their respective sources. A graphic analysis of the spectrum of a bomber treated with sound-absorbing material is shown in the top half of Figure 1. Since a filter of a bandwidth of 5 c effect of low-frequency noise components (below about 200 c) in voice communications in a wide-band noise is, in many cases, comparatively small. It has been found preferable in most cases to use a filter set variable in convenient discrete steps. A filter set covering the range from about 40 to 9,000 c in eight octaves has been found adequate and useful. The analysis of the noise spectrum mentioned above with

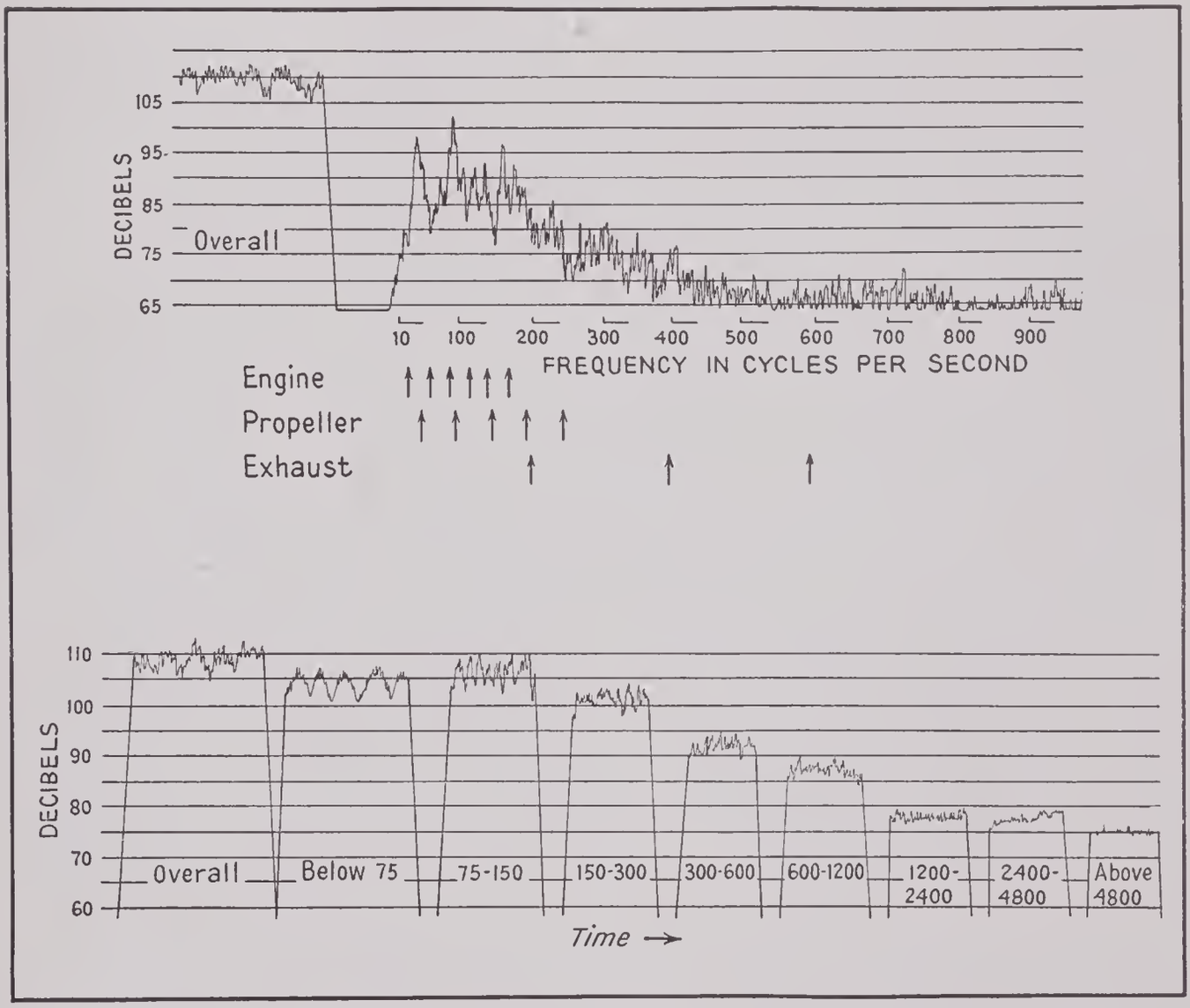


FIGURE 1. Noise in multi-engine bomber, treated with absorbing material. Top—analysis with narrow band-pass filter. Bottom—analysis in octave bands.

has been used, enough resolution was obtained to distinguish the noise components produced by the engine, propeller, and exhaust.

In many instances the type of analysis mentioned above is too time-consuming. Furthermore, the detailed information afforded in the low frequencies is often not necessary, since the the octave filter set is shown in the lower part of Figure 1. An "overall" position, with no filters in the circuit, is added.

The indicating instrument is of the square-law type and is calibrated in decibels. A meter with pointer indication and a graphic level recorder are provided. The recorder is used more

frequently in field work where automatic operation is desirable.

An automatic octave sound analyzer and recorder designed by the Electro-Acoustic Laboratory is shown in Figure 2. This system is composed of the following parts:

1. The microphone.
2. The octave band-pass filter set.
3. The sound-level meter (containing the amplifiers and the indicating meter).

up the required flight conditions. Pressure on the pushbutton of the control unit sets the device in operation. For each of the nine filter positions in succession, the sound level is recorded automatically by the level recorder on waxed paper tape. The filter bands are changed automatically by a relay. After all nine bands have been used, the device is ready for another analysis which is indicated to the pilot by a light signal.

FIGURE 2. Automatic octave sound analyzer with recorder.

4. The graphic level recorder.
5. The hand-held remote control unit.
6. The battery power supply.

The equipment was specially designed for automatic operation, portability, and compactness. It is a matter of relative ease to install it in the rear compartment of a single-seater airplane and place the control unit in the hands of the pilot. After the microphone has been placed at the desired position and after adjustment and calibration of the apparatus, the pilot can set

The case shown at the left in Figure 2 contains the filter set, sound-level meter and batteries, and weighs about 125 lb. Its dimensions are 19 x 18 x 11 in. The second case contains the recorder and the rest of the batteries. It weighs about 105 lb and has the dimensions 19 x 18 x 12 in. In case space is very limited, the units can be broken down into their component parts, as shown in Figure 3. This equipment is described in detail in a report[12] issued by the Electro-Acoustic Laboratory. That laboratory has also

assisted the Services in writing the Army-Navy Specifications AN-N-7 for noise-measuring equipment, in which further details may be found.

The data presented in the following sections were taken using equipment of the type described.

2.1.2 Noise in Airplanes

The most prominent low-frequency components in the noise spectrum of airplanes are directly related to the propeller-tip passage frequency and its harmonics. The noise of the continuous type at medium and high frequencies is due to the aerodynamic noise of the air rushing past the fuselage and to turbulence noise produced largely at the propeller tips. The noise spectrum and levels inside an aircraft cabin depend *primarily* upon the following factors:

1. Engine power.
2. Propeller-tip speed.
3. Minimum separation of propeller tips from fuselage.
4. Exhaust type.
5. Airspeed.
6. Type of airplane and location of cabin.
7. Transmissive and absorptive properties of the cabin walls.
8. The quality of seals around windows, hatches, and the type of cabin ventilation used.

Noise analyses have been carried out by the Electro-Acoustic Laboratory in more than thirty-five different types of military aircraft under a variety of flight conditions. A number of reports[18a, 21, 25, 31] have been issued on the subject.

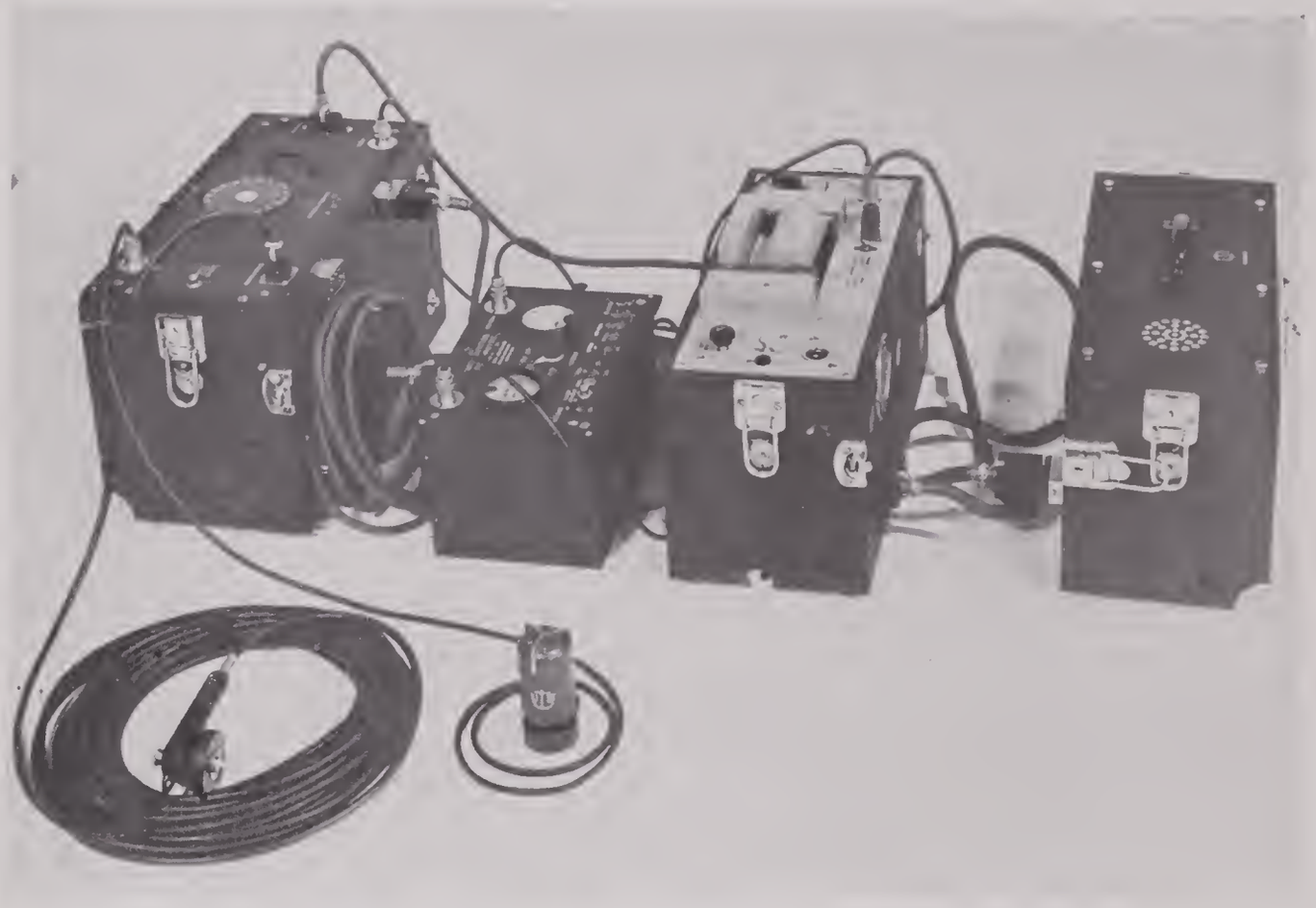


FIGURE 3. Sound analyzer—component parts.

Sample spectra for a number of aircraft of different types are shown in Figures 4 and 5. It is evident that, in modern, high-speed, high-powered, single-engine aircraft, noise conditions are more severe than in large multi-engine bombing craft. As a rule, only multi-engine

airplanes afford space and opportunity for installation of sound-absorbing treatment. This can materially reduce the sound levels at medium and high frequencies and results in improvement of voice communications.

Note that in the jet-propelled airplane (see Figure 5) the absence of a propeller and conventional engine results in low sound levels in the low-frequency bands. All spectra shown are for airplanes which are not treated by installation of sound-absorbing material.

Figure 6 shows how the noise level and spectrum changes with varying engine power conditions. As the propeller-tip velocity reaches high values, the high-frequency components of the spectrum increase more and more relative to the low-frequency components. An analysis with a narrow-band analyzer shows that the harmonics of the propeller-tip passage frequency increase very rapidly under those conditions.

As a further illustration of the many variables entering into the picture, consider the influence of altitude on the noise levels and

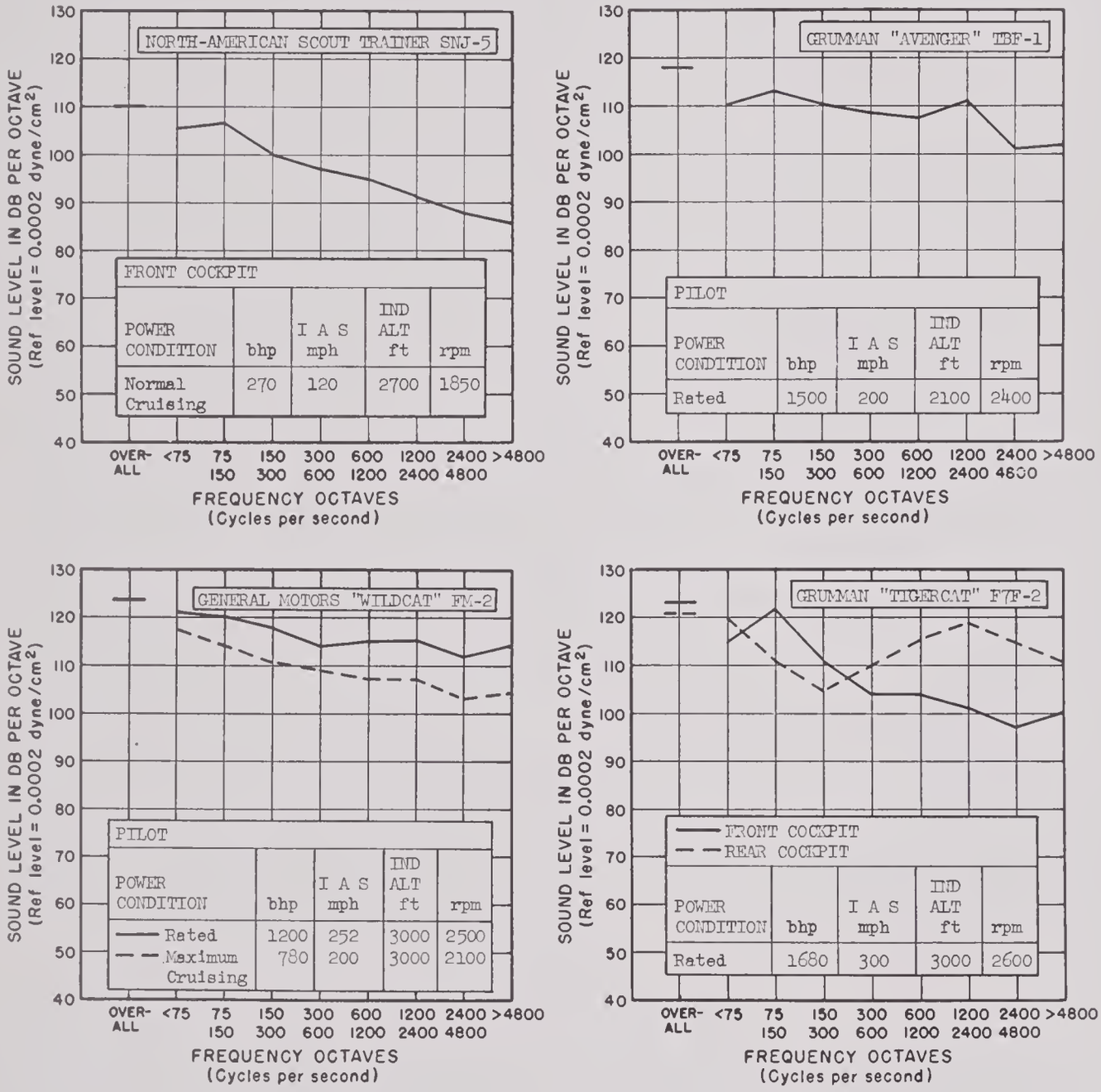


FIGURE 4. Noise levels and spectra in aircraft.

spectra. The differences in the sound levels measured due to a change in altitude in octave bands are plotted for a P-47D and a B-17F airplane for constant engine power in Figures 7 and 8. The changes in spectrum and level are slight even for altitudes near the ceilings of the aircraft involved.

is shown by the contours for a PBM-3 airplane in Figure 10.

To obtain an estimate of the sound levels generated by the firing of machine guns in a typical bomber, such as the B-17F, the microphone was placed at the top turret gunner's knees. A reproduction of the record obtained during gun-

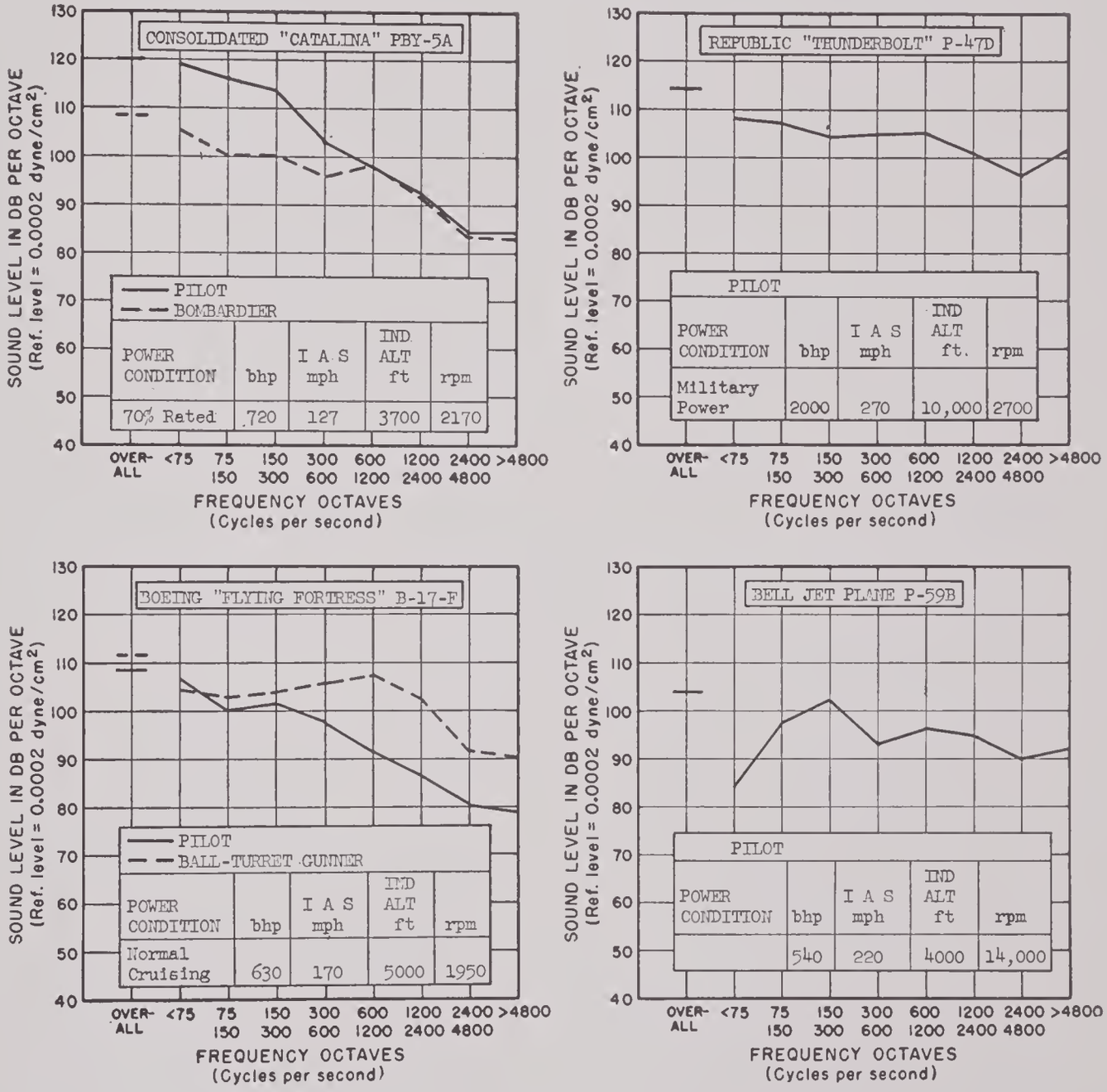


FIGURE 5. Noise levels and spectra in aircraft.

The variations of the levels inside a cabin are illustrated by the contours of equal sound levels for a DC-3 transport airplane, given in Figure 9.

How an imperfectly sealed window can increase the levels in certain bands in its vicinity

nery practice is shown in Figure 11. Instantaneous increases in the overall sound level of 10 db or more were noticed.

The noise levels at high frequencies can be materially reduced by proper installation of

sound-absorbing material. The added weight can be kept within comparatively small limits. Figure 12 gives an estimate of the reduction of noise levels which was realized by the experi-

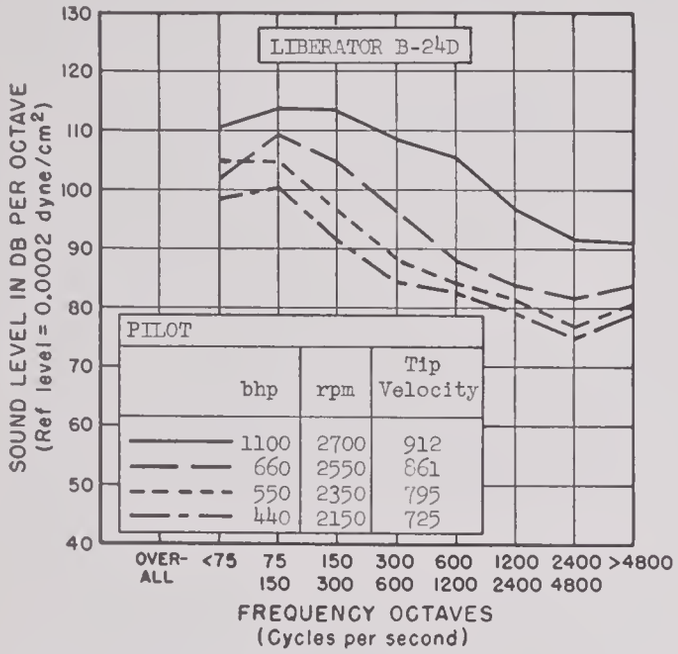


FIGURE 6. Noise levels in a B-24D bomber as a function of engine power.

mental installation of a small amount (about 40 lb) of additional absorbing material in a Mitchell type B-25C bomber. Although the lower octave levels and the overall levels are, in general, not markedly affected by the treatment, the reduction of the levels in the upper octaves may yield marked improvement in the performance of the interphone aboard. The proper design and installation of acoustical treatment is discussed in Section 2.3.

2.1.3 Noise in Tanks and Landing Vehicles

In Figure 13 noise spectra and levels measured inside the turret of a medium and light tank are given. Among the chief sources of noise are the clatter of sprockets and treads and the engine. Although the noise measurements discussed here were made early in World War II, the spectra found in tanks of later design are not likely to be very different in character. There is every reason to believe, however, that the levels will be at least as high as shown in Figure 13, if not higher. Increased engine power, speed, and fire power all make for increased noise. Judicious application of sound-absorbing materials, vibration insulation of critical parts, and replacement of steel treads with rubber treads can markedly reduce the noise levels in the important bands. As an example, the noise in a Navy Amphibious Landing Vehicle (LVT-4)[24] is analyzed and measures for its reduction are described below. It should be emphasized that proper preproduction acoustic engineering and design is preferable to any postproduction modifications.

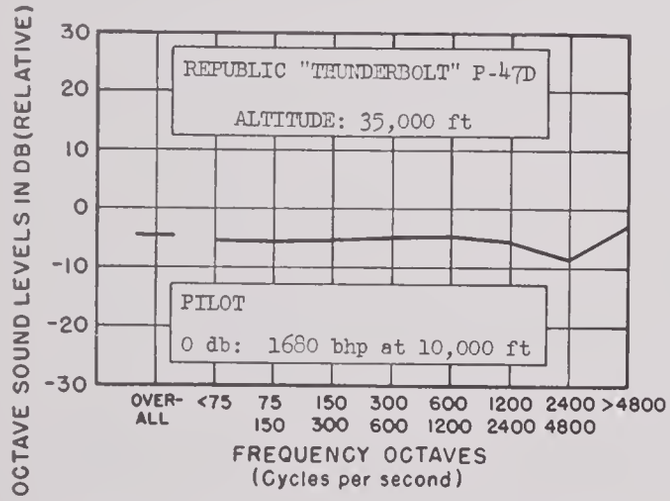


FIGURE 7. Effect of altitude on the sound levels in a P-47D airplane for constant engine power.

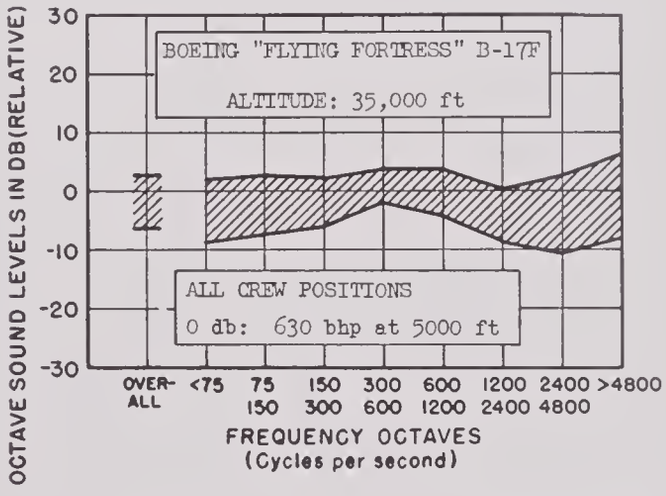


FIGURE 8. Effect of altitude on the sound levels in a B-17F airplane for constant engine power.

Figure 14 shows the noise levels and spectra in the (untreated) cab of an LVT-4. The levels are among the highest found in any combat vehicle. Voice communication under these con-

ditions is very difficult and temporary deafness of combat personnel is common.

The sources of noise contributing to the levels

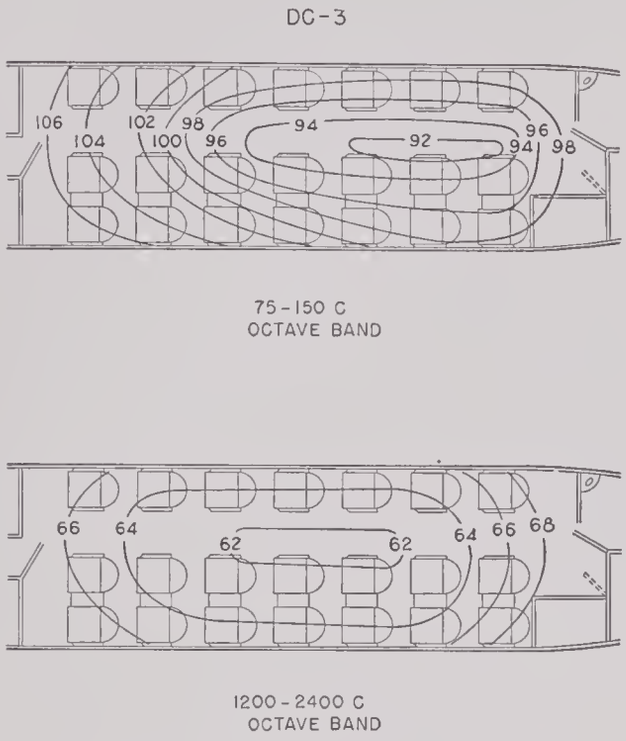


FIGURE 9. Contours of equal sound levels in a DC-3 airplane.

in the important bands are, in order of decreasing importance,

1. Forward sprocket drive.
2. Engine.

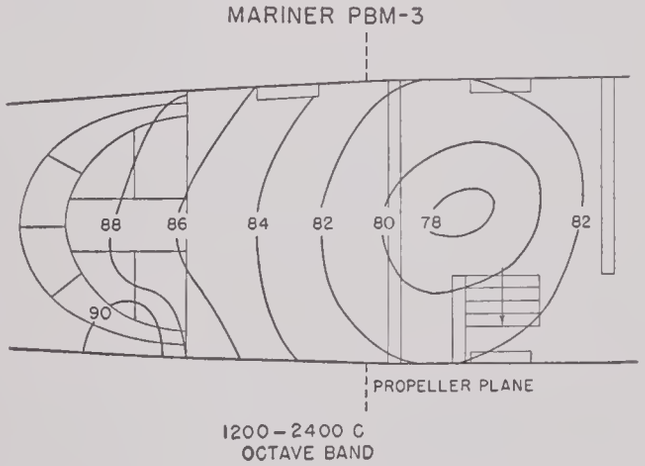


FIGURE 10. Contours of equal sound levels in a PBM-3 airplane.

3. Transmission.
4. Chain, rear sprocket, exhaust, etc.

A cross section through the vehicle is shown in Figure 15, with the location of the noise sources indicated in detail. In passing, it may be noted that in the present case the exhaust constitutes the chief contributor of *low-frequency*

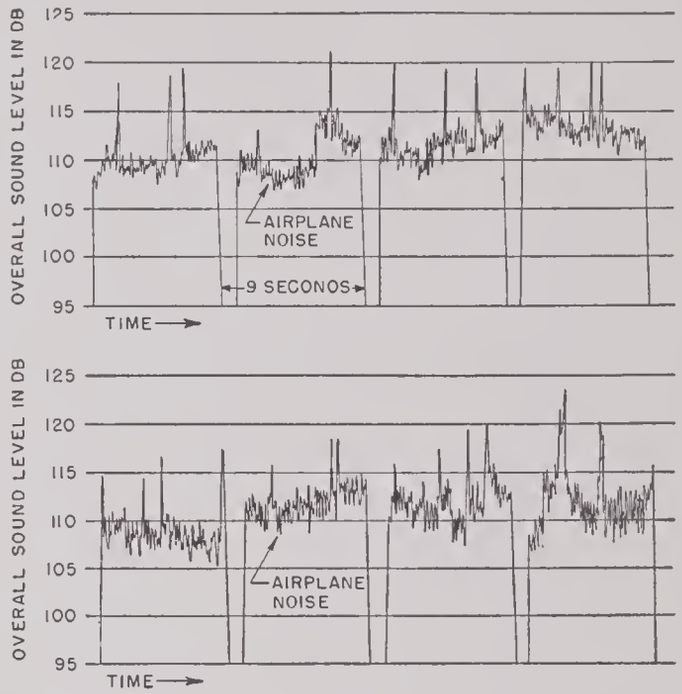


FIGURE 11. Transient noise levels due to firing of two machine guns in a B-17F airplane.

noise. Introduction of a suitable muffler results in considerable improvement.

By application of sound-absorbing material

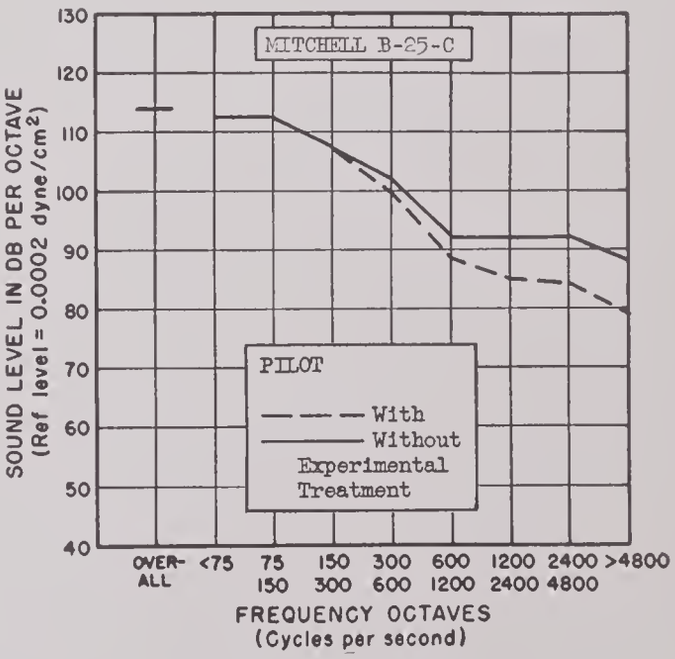


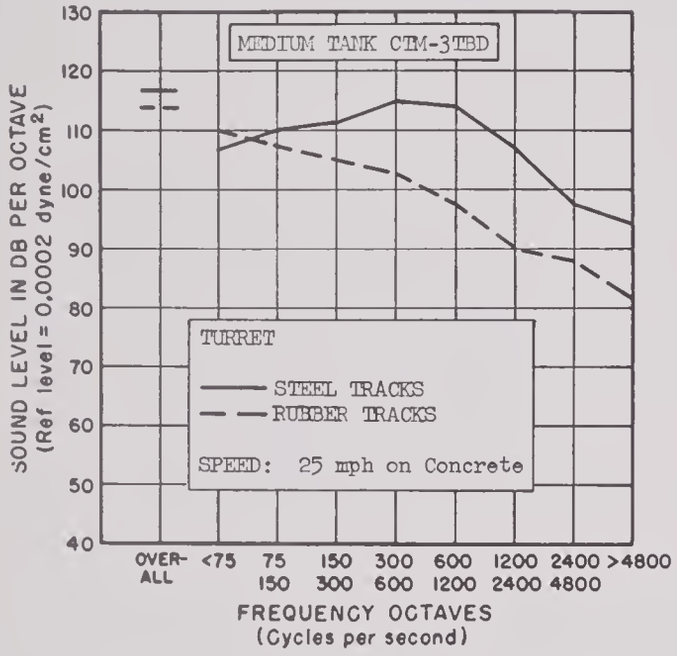
FIGURE 12. Effect of experimental installation of sound-absorbing material in a B-25C airplane.

to the interior of the cab and by replacing the forward-drive sprockets with sprockets having

rubber vibration insulators, the noise can be reduced. Figure 14 shows that reductions in noise level of the order of 10 db are realized.

2.1.4 Noise on Ships and Boats

The sources of noise on warships are of a varied nature, especially at exposed stations.



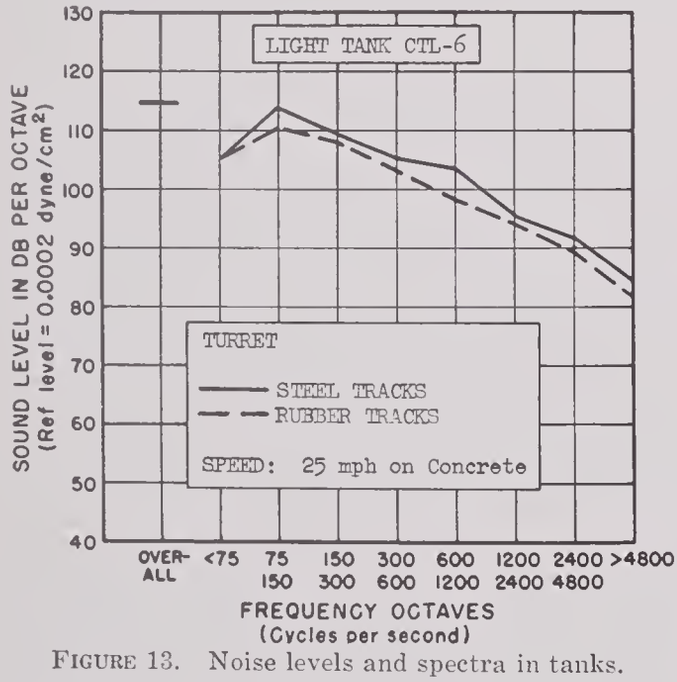


FIGURE 13. Noise levels and spectra in tanks.

Gunfire, airplane noise on aircraft carriers, and wind are among the most important ones. There are, however, also important stations below deck where the ambient noise is considerable. As an example, the noise spectrum in the engine room of a submarine is shown in Figure 16. The levels are considerable and communication is seriously interfered with.

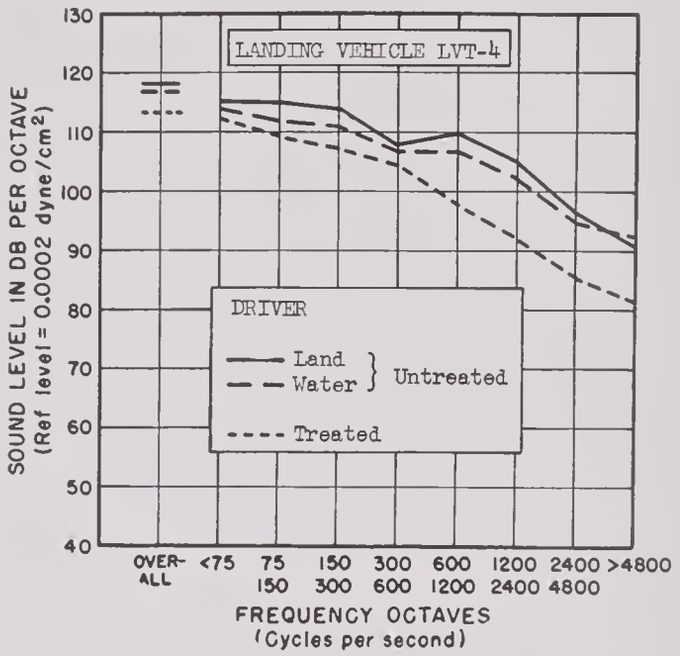


FIGURE 14. Noise levels and spectra in an LVT-4 landing vehicle.

In certain situations in World War II it was necessary to land military personnel in small boats equipped with outboard motors on enemy-held shores. It was of vital importance to escape detection. In order to evaluate quantitatively

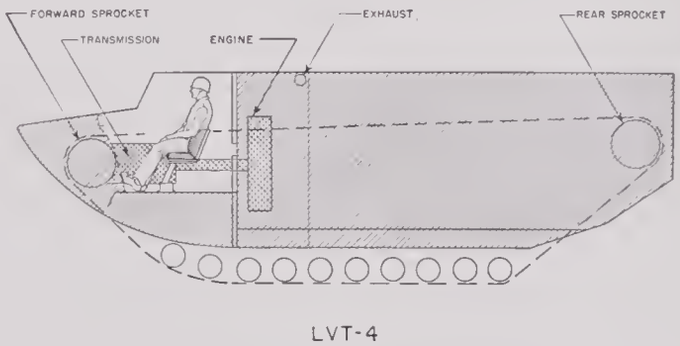


FIGURE 15. Main sources of noise in an LVT-4 landing vehicle.

the sound levels produced by a typical outboard motor (10 to 20 hp) measurements were made[30] by the Electro-Acoustic Laboratory. A highly directional tubular microphone was used which was mounted on shore. The boat was held stationary at a distance of about 45 ft. (See Figure 17.) By rotating the boat, a directional diagram

of the sound source could be obtained. A dummy propeller was used and the motor adjusted to rated power conditions. Figure 18 shows the directional pattern of the motor noise. It appears from the diagram that the radiation in

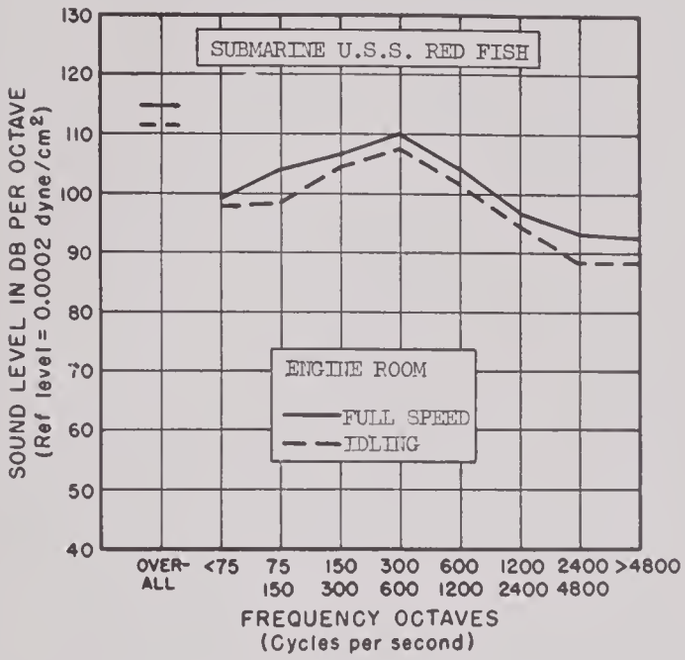


FIGURE 16. Typical noise levels and spectra in the engine room of a submarine.

the direction of the bow is comparatively weak. Radiation is strongest in a direction perpendicular to the boat's course. The solid line in Figure 19 shows the noise spectrum of a typical outboard motor. The dashed line was obtained after

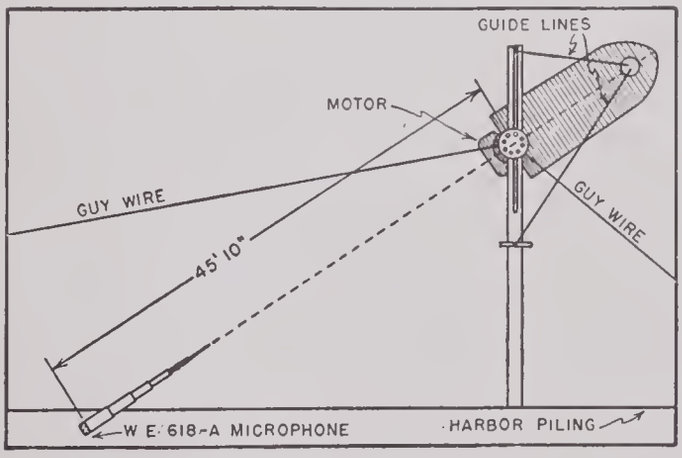


FIGURE 17. Experimental determination of directional characteristics of outboard motor noise.

installation of a jacket of sound-absorbing material and suitable intake and exhaust silencers. Jury tests of the maximum detectable distance of the approaching boat confirm the effectiveness of treatment. This distance is about 300 to 500 yd at full speed but depends strongly on the prevailing wind, among other things.

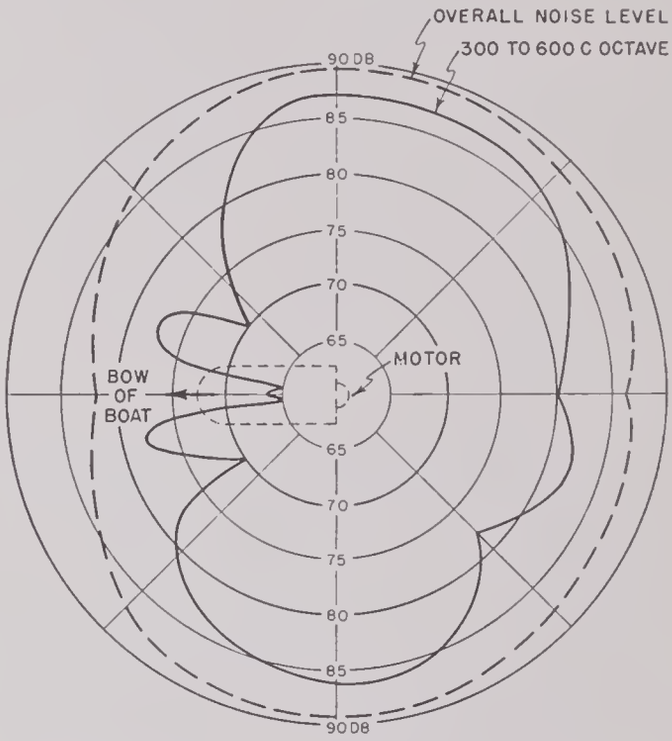


FIGURE 18. Directional characteristics of outboard motor noise.

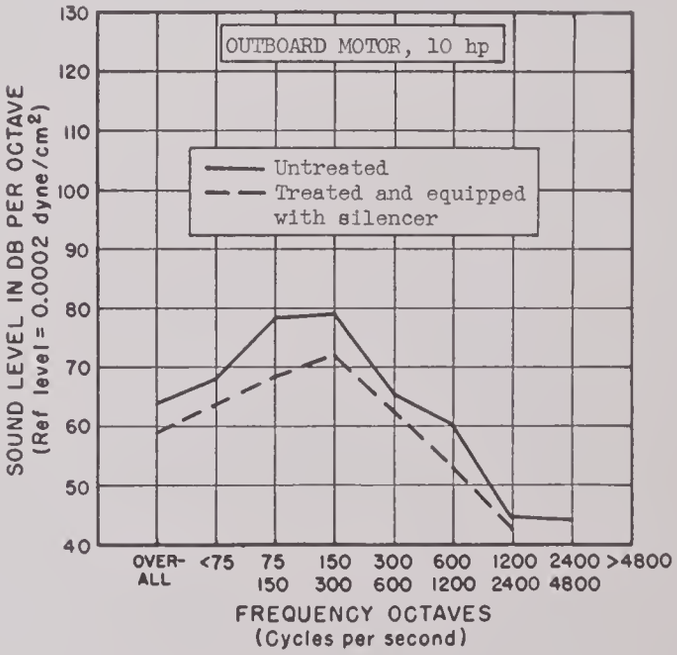


FIGURE 19. Noise levels and spectra of a typical small outboard motor.

2.1.5 Miscellaneous

In this section miscellaneous sound-level measurements are discussed which were carried

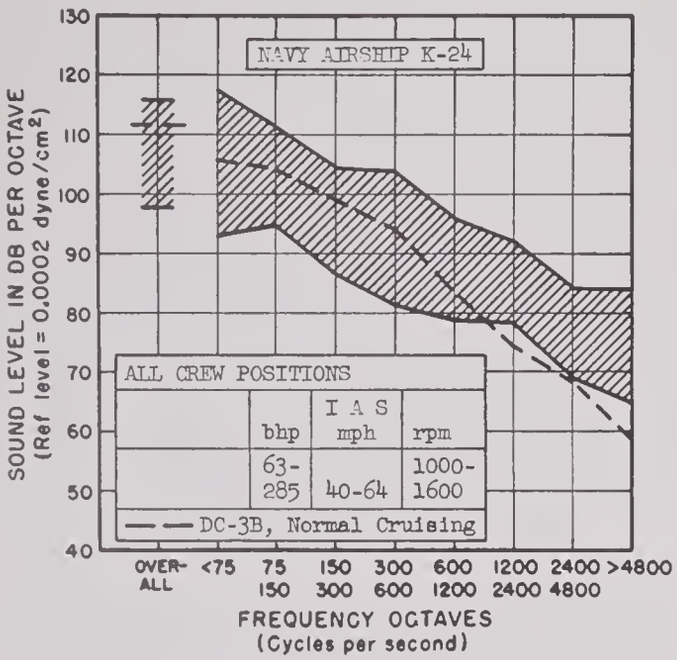


FIGURE 20. Comparison of noise spectra in a type K airship and a DC-3B airplane.

out by the Electro-Acoustic Laboratory and which are of more than passing interest.

Sound levels and spectra in *Navy Type K airships* have been investigated extensively and the findings were published in a report.[29] The sound levels in the cabin are comparatively low, even without sound-absorbing material. Figure 20 shows the envelope for all crew positions for a variety of flight conditions. For comparison, the spectrum for a DC-3B transport plane has been added in the figure. The noise conditions in the two ships are, on the average, very nearly alike.

Figure 21 shows a section through the airship car with some of the crew positions indicated. It is of interest to study the change in sound levels as the test microphone is carried along the cab as a function of the distance from the propeller plane. The results of such measurements for two constant power conditions are

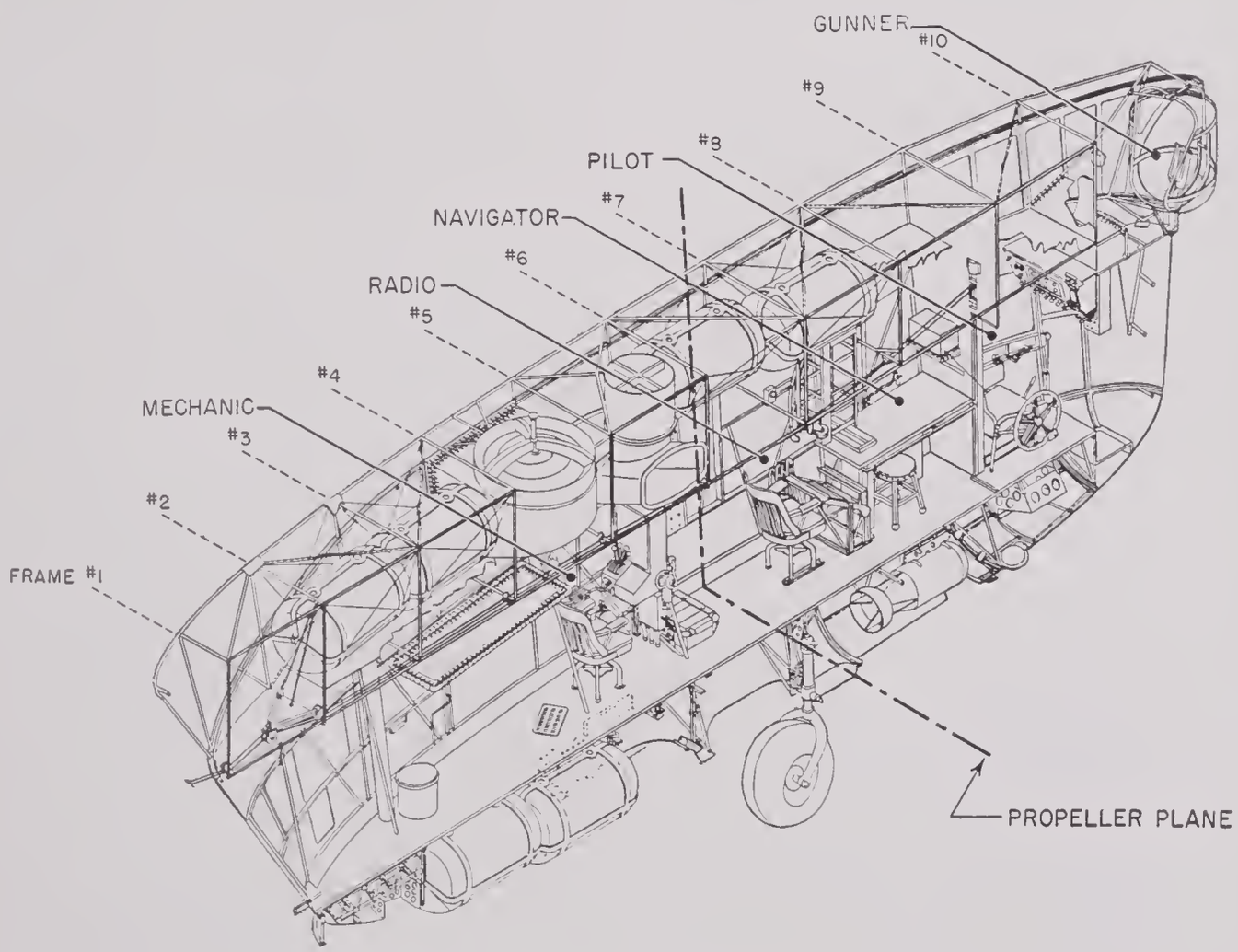


FIGURE 21. Crew positions in cab of Type K airship.

plotted in Figure 22. The ordinate is the difference, in decibels, of the average levels in all bands for a given distance from the propeller plane and the average levels measured in the propeller plane.

The sound levels due to an airplane passing overhead are of interest in certain questions of

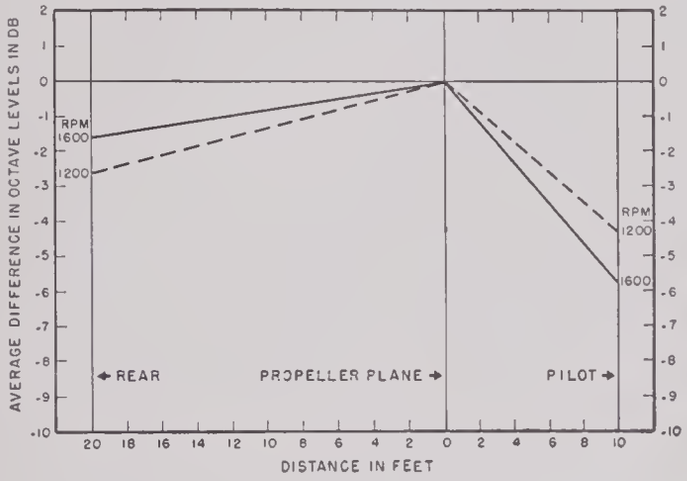


FIGURE 22. Variation of noise levels with distance from propeller plane in cab of Type K airship.

countermeasures against low-level attack by aircraft. The Electro-Acoustic Laboratory carried out a series of measurements of the sound levels at ground level due to a medium bomber flying overhead in level flight at a specified altitude. A B-18 medium bomber was made available for this purpose.

A dynamic microphone equipped with a windscreen was placed on the ground at the airport where the tests were performed.[15] It was connected through suitable amplifiers to a Miller film recorder. The pilot was instructed to fly the airplane in a straight course over the microphone location, maintaining a constant altitude and specified flight conditions.

The sound picked up by the microphone was recorded as a function of time for various altitudes and flight conditions. A reference time mark was introduced by shorting out the signal when the craft was directly overhead.

The Miller film recording method has been described in the literature.[1] The recording medium is a narrow film tape, consisting of a celluloid base and a layer of transparent soft gelatin covered with a thin, uniform layer of a chemical opaque to light. The recording head consists of a wedge-shaped stylus driven by an electromagnetic drive. The stylus removes the opaque layer in varying widths, corresponding to the wave form of the signal. Reproduction is effected by conventional optical means. Figure 23 shows a sound track for airplane noise. A 60-c wave has been recorded for comparison.

The records obtained in the field were played back and analyzed in the laboratory. Two methods were used.

1. The playback signal was analyzed by a variable band-pass filter whose output signal was recorded by means of a high-speed-level recorder. By successive playbacks, the levels in each octave band were recorded as a function

FIGURE 23. Recording of airplane noise on Miller film.

of time. Figure 24 shows a set of such records for maximum power conditions and an altitude A of 100 ft. The dip in each record corresponds to the time mark indicating the airplane to be overhead. H is the horizontal distance of the airplane from the vertical through the microphone location. For convenience, the *maximum* levels in each band have been plotted in Figure 25, with altitude as parameter. It is evident from Figure 24 that in the first octave the maximum sound level is attained after the airplane has passed the microphone location by several hundred feet. This points to the well-known existence of maxima in the radiation pattern near the propeller plane.

2. A more detailed frequency analysis was afforded by the use of a narrow-band (20 c wide) wave analyzer. The film containing the record of one flight was pasted together to form a loop which made repeated playback con-

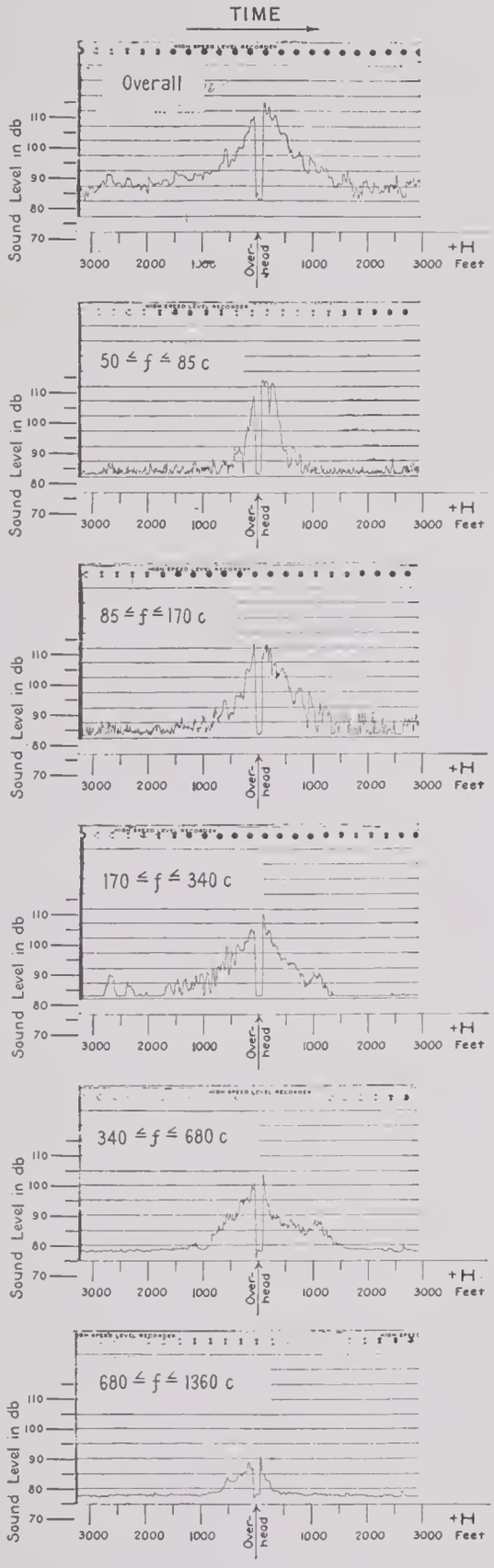


FIGURE 24. Sound levels due to an airplane passing overhead as a function of horizontal distance H.

venient. For each center frequency of the filter pass band the maximum sound level was read on the analyzer and plotted as a function of frequency in Figure 26. Good correlation of the peaks in the spectrum with the propeller-tip passage frequency and its harmonics was obtained.

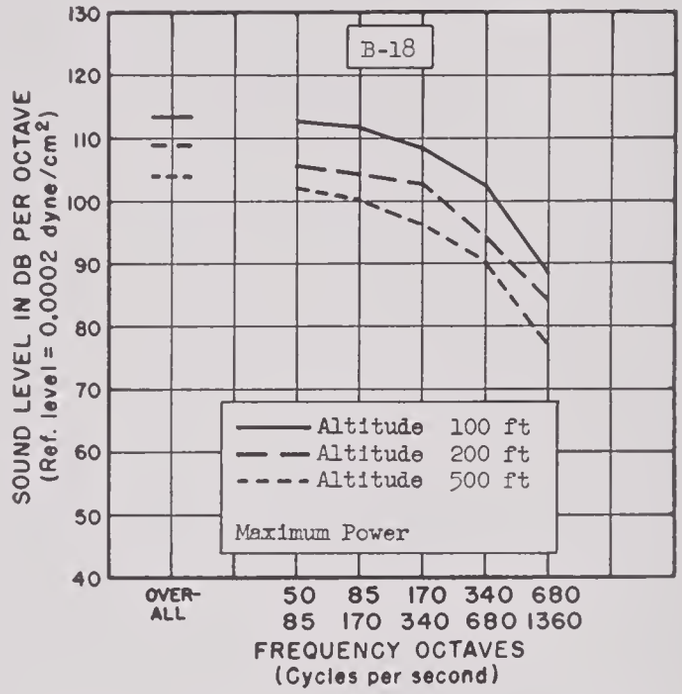


FIGURE 25. Maximum sound levels due to a B-18 bomber passing overhead.

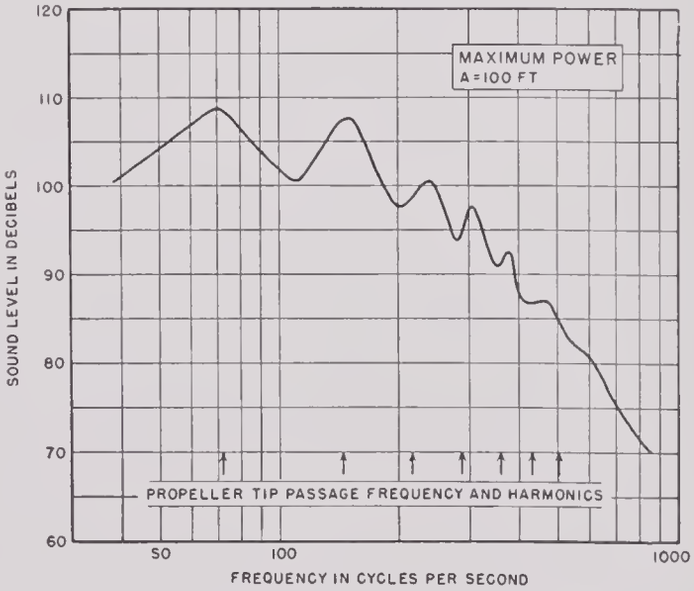


FIGURE 26. Maximum sound levels in 20 c bands due to a B-18 bomber passing overhead (altitude 100 ft).

For more accurate work doppler corrections and corrections due to the finite sonic velocity would have to be applied.[1]

In *aircraft factories* and similar establishments noise levels may be so severe as to lead to temporary and, after long exposures, permanent hearing losses of the personnel involved. Figure 27 gives representative noise spectra and levels near various noise sources in an aircraft factory.[14] Many noises encountered there are of a transient, impulsive nature and the levels vary considerably as a function of time. In such instances the cross-hatched areas in Figure 27 are an attempt to indicate the range of the fluctuations. The noise levels are seen to be particularly severe at high frequencies, where hearing loss is most easily incurred. Effective aural protection for the operating personnel is a necessity under such conditions.

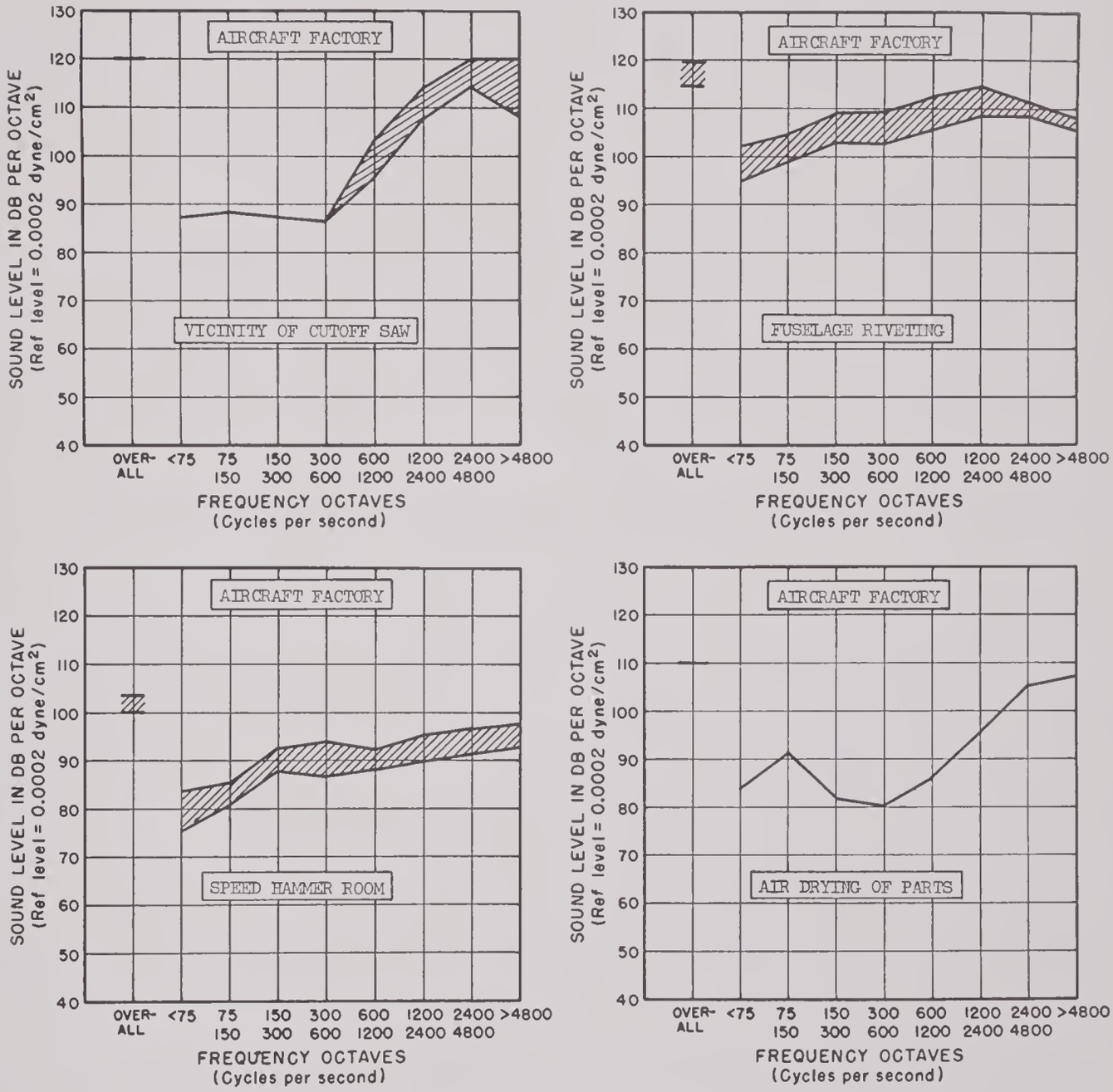


FIGURE 27. Noise levels and spectra at various stations in an aircraft factory.

2.2 REDUCTION OF AMBIENT NOISE AND VIBRATION

2.2.1 General Considerations

A typical section of the fuselage in an airplane which is treated with absorbent material

consists of the following two parts: the outer skin, made of Dural, and the structure containing the acoustic material, separated from the skin in most cases by an air space an inch or two in depth. Sound originating outside the airplane and striking the Dural skin will be partially *transmitted* into the interior of the cabin. A sound field will be set up in the interior of the cabin which will, in part, be determined by the fact that some of the sound inside the cabin will be *absorbed* by the acoustic material. Although this description is somewhat crude and the physical processes of transmission and absorption are not easily separable, it has been found helpful, nevertheless, to separate them in practice and to define a *transmission coefficient* τ and an *absorption coefficient* α. The transmission coefficient τ is defined as the ratio of the sound energy transmitted *through* the acoustic structure to that incident upon it. It is often more convenient to express τ in terms of the *attenuation* N in decibels, according to the following relation

$$\tau = 10^{-N/10}. \qquad (1)$$

The absorption coefficient α is defined as the ratio of the sound energy absorbed by the acoustic structure in question to that incident upon it from the cabin inside. Although these definitions are useful for the purpose of discussion, a meaningful experimental evaluation of τ and α will require restatement and modifications. This is discussed in Section 2.2.3.

To elaborate somewhat on the general picture, consider an enclosure, say an airplane cabin, immersed in a sound field of uniform sound energy density. Assume further that the walls of the cabin are uniform and that the structure has a transmission coefficient τ and an absorption coefficient α. If it is assumed that the resultant sound field inside the cabin is also of uniform sound energy density, it can be shown that the reduction in sound level due to the walls equals 10 log $(\tau + \alpha)/\tau$, in decibels. A plot of this function is shown in Figure 28. Contours of constant reduction in sound level are plotted as a function of α and τ. The chart is not applicable to practical cases except in broad terms, due to oversimplification of the underlying assumptions. It does, however, reveal the important fact that appreciable values of sound-level reduction are possible only if α is appreciable *and* τ is small.

Clearly, it is impractical to determine the sound-level reduction by direct experimental means in an aircraft. Considerable research has

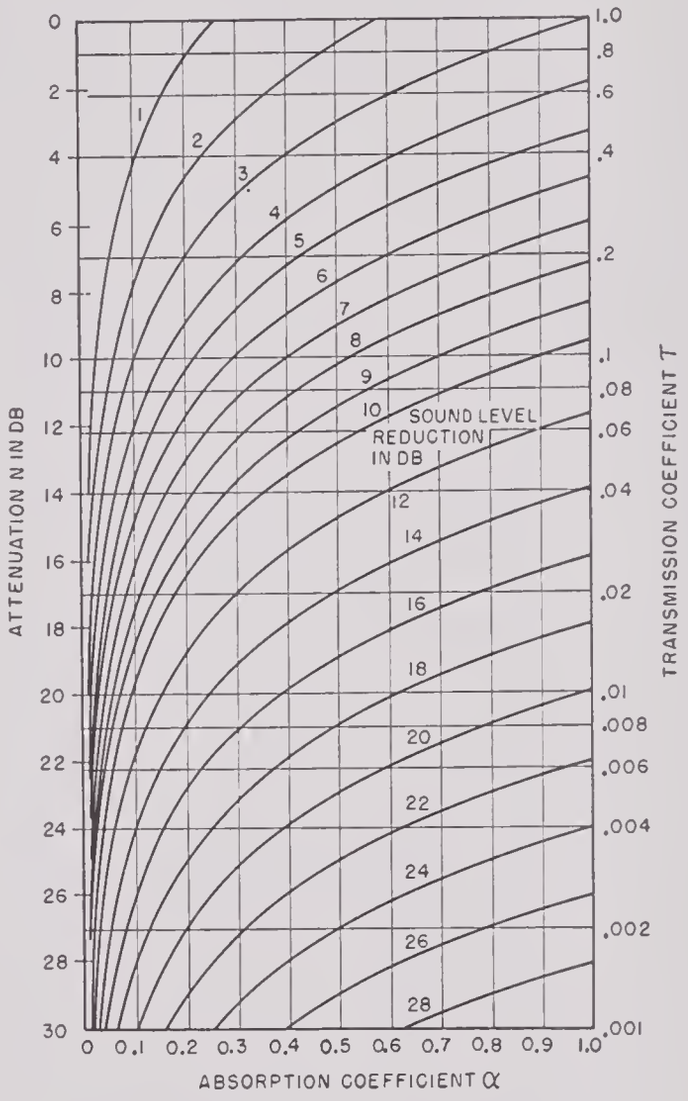


FIGURE 28. Theoretical reduction of sound levels in a cabin whose walls have a transmission coefficient τ and an absorption coefficient α.

shown that laboratory measurements under certain specified conditions yield values *related* to the absorption coefficient α and transmission coefficient τ, as defined above, which have been successfully used in predicting the actual performance of a structure.

In the following, α and N (and τ) will be used *only* to denote quantities obtained by a specified apparatus in a specified manner in the labora-

tory.[b] For the details of procedure and apparatus, Section 2.2.3 should be consulted.

The process of sound absorption in porous materials can be described roughly as follows: Sound waves impinging on the material force air in and out of the channels between the fibers of the material, where energy dissipation due to friction takes place. In materials not containing an impervious septum, air can be forced through at a steady flow. It can be shown that the resistance to steady flow, the so-called *flow resistance* R, of a material can be correlated with its absorptive properties.

Weight, thickness, and the *space configuration* of the components of a structure are other important parameters in determining the effectiveness of a structure as an acoustical barrier to transmitted sound and an absorber of sound inside the enclosure whose walls are formed by that structure. There are, for a given structure, optimum combinations of these parameters.

The emphasis placed on aircraft in this and the following sections is, for the most part, due to the relative importance of aviation in World War II. It is not intended to imply (1) that noise conditions are unimportant in other combat vehicles (see Section 2.1.3) and (2) that the results of the research and methods for noise prediction and reduction are not applicable to cases other than aircraft.

Anticipating the results of later sections, it is the medium- and high-frequency noise components which seriously interfere with communications. In view of this fact, it is in general not worthwhile to try to reduce the low-frequency vibrations of windows, panels, etc., in aircraft. Their action as secondary sound sources at low frequencies may result in some increase of the sound levels at certain low frequencies. It is, in general, more efficient to forego special vibration treatment of such panels and windows in favor of design for maximum noise reduction for medium and high frequencies.

In conclusion, a word should be said with reference to the problem of *vibration* to which military personnel is exposed in aircraft and other combat vehicles. Reduction of excessive vibrations sometimes encountered at certain crew positions can be effected by more or less conventional techniques. An example of vibration insulation of a pilot's seat is given in a report[5] compiled by the Electro-Acoustic Laboratory.

Vibrations affecting structural design are beyond the scope of this discussion.

[b] This is not strictly true with regard to α, since an estimate of the values obtained by two different experimental techniques was found most useful.

2.2.2 Properties of Acoustic Materials and Structures

The general requirements which an acoustic material for sound control in combat vehicles must meet are as follows.

1. It must attenuate sound effectively in the important frequency range.
2. It must absorb sound effectively in the important frequency range.
3. It must be light in weight.
4. It must have low thermal conductivity.
5. It must have high flame resistance.
6. It must have high corrosion resistance.
7. It must have adequate mechanical strength.
8. It must be vibration-proof.
9. It must be moisture resistant and quick-drying.
10. It must resist rot and fungi.
11. It must be vermin-proof.

Sound Attenuation

The simplest structure consists of a single *impervious septum*, say, the outer Dural skin of an airplane. At low frequencies it is possible to calculate the attenuation N of such a structure from the physical properties of the septum and the mounting conditions. In the vicinity of the normal modes of vibration N is found to be very small. Between resonances, the attenuation is little greater than the values resulting from the assumption that the panel can be replaced by a pure mass with zero stiffness. At frequencies above several hundred cycles per second the *average* attenuation can be predicted from the surface density σ of the panel alone.[18] If uniform sound-pressure distribution over one side of the

panel is assumed and if it is further assumed that the panel is terminated on the other side by a perfectly absorbing termination,

$$N \approx 20 \log \frac{\omega\sigma}{R_0} \qquad (2)$$

if N is identified with the ratio, in decibels, of the sound pressures in front and in back of the panel.

σ = Surface density of panel in grams per square centimeter.
$\omega = 2\pi f$ (f, frequency in cycles).
R_0 = 42 grams per square centimeter per second.

The expression R_0 is the specific acoustic impedance of the perfectly absorbing termination, equivalent to free air. Equation (2) in this connection is often referred to as the *weight law*.

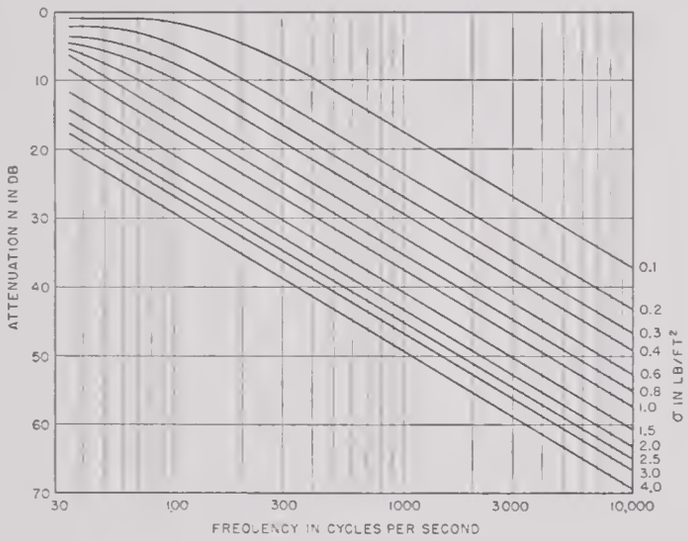


FIGURE 29. Theoretical values of attenuation N vs frequency for weight law.

Figure 29 shows computed values of the attenuation N, in decibels, for various values of surface density σ.

Figure 30 shows an experimental curve of N vs frequency for a Plexiglas window obtained on the attenuation measuring apparatus to be described in Section 2.2.3. The appropriate weight law curve is drawn in, for comparison, from Figure 29. Note the prominent individual resonances, especially at low frequencies. The fluctuations and irregularities are considerable everywhere. But it is clear from inspection that the weight law curve seems to approximate the average value of N at medium and high frequencies reasonably well. The weight law will be used as a practical yardstick of comparison of the attenuating properties of structures.

As an example of such a structure, consider

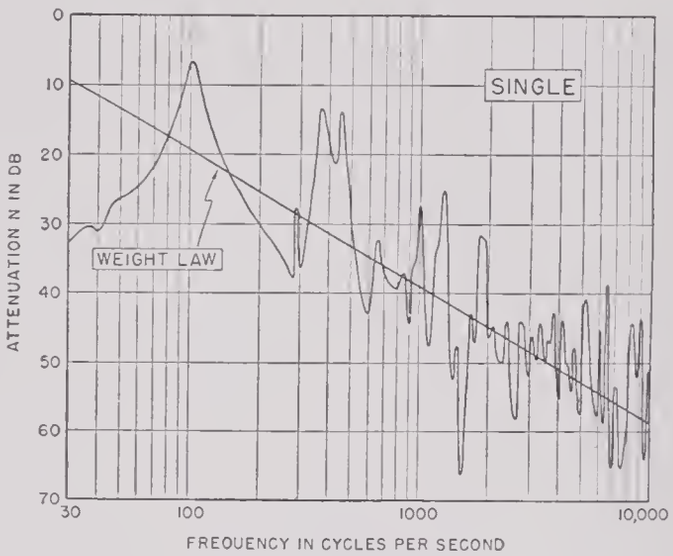


FIGURE 30. Attenuation of a single Plexiglas window.

Figure 31, on which N for a double impervious septum (double Plexiglas window) is plotted vs frequency. This structure is considerably "better" than the weight law.

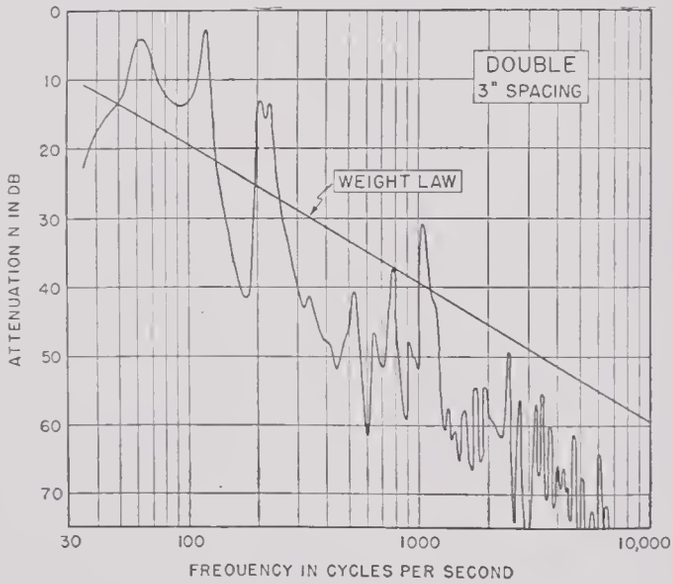


FIGURE 31. Attenuation of a double Plexiglas window.

From investigations of various structures consisting *only* of impervious septa the following conclusions can be drawn with respect to their attenuation properties (sound absorption by these structures is very small):

1. Impervious septa are, at low frequencies, almost transparent to sound. The low-frequency resonances can be damped by cementing or spraying suitable damping materials on the septum. The practical value of such treatment is controversial, since the transmission of sound at medium and high frequencies is not appreciably affected.

2. The average attenuation at medium and high frequencies of single septa (of sufficient area) can be predicted from the weight law. The controlling factor is the surface density. Mounting conditions, size, shape, etc., have negligible influence.

3. Double septa attenuate sound better than the weight law would predict at medium and high frequencies. At low frequencies they afford no appreciable advantage. In practice, some advantage may be lost owing to the necessity of using heavier frames for the double septum. Comparisons should be made on the basis of equal total weight.

4. There is no advantage in supporting windows in elastic mountings.

Absorptive structures consist of at least one layer of sound-absorbing material and one or more impervious septa. A large number of structures are discussed in detail in Report OSRD 1543.[18] Equivalent electric circuits can be drawn which describe the general behavior of structures at low frequencies. Numerous examples are given in the report referenced above.[18] The attenuation vs frequency characteristics of single and multiple absorptive structures are similar in general character to the characteristics shown in Figure 31. The average attenuation values at medium and high frequencies are always greater than the values predicted by the weight law. Tests have shown that among the many possible structures the three configurations shown in Figure 32 are among the most effective. Structure V is the configuration recommended in the Army-Navy Aeronautical Specifications for Sound-proofing Materials AN-S-32$_a$ and AN-S-33$_a$.

It is desirable to have a measure of the effectiveness of structures with respect to their ability to attenuate sound. It has been suggested to take the amount by which the average attenuation of the structure exceeds the attenuation predicted by the weight law at a frequency of 5,000 c. To take into account the weight of different materials, the difference in attenuation,

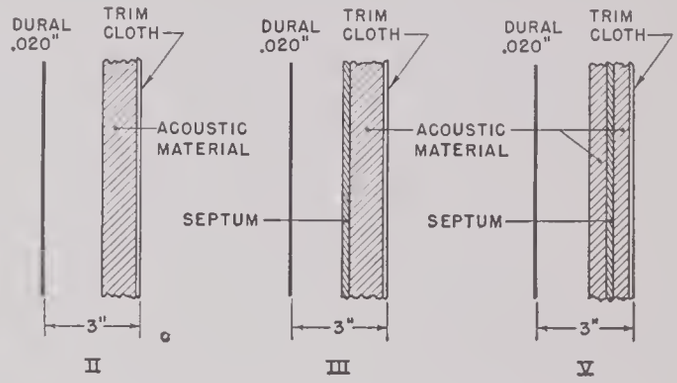


FIGURE 32. Effective single absorptive structures.

n_5, is divided by σ_s, the surface density of the total structure *exclusive* of the *Dural* skin. A *merit factor* is therefore defined as follows:

$$\text{Merit Factor [MF]} = \frac{n_5}{\sigma_s}. \qquad (3)$$

The usefulness of the MF as a basis of comparison between different *structures* lies in the fact that it affords a *rank-ordering* of the effectiveness of the structures to attenuate sound per unit weight. A quantitative comparison other than rank-ordering is in general not possible. It should be remembered, however, that in addition the weight of the structure has always to be considered.

Acoustic materials per se may be rank-ordered by use of the MF when placed in a given structure. Tests have shown that Structure III (see Figure 32) has a reasonably constant MF for a given acoustic material independent of its *thickness T*. For this structure, the MF ranges between about 15 and 135 for various acoustic materials.

A detailed discussion relating merit factor and the physical characteristics of the material and the attenuating properties of structures containing this material is given in Report OSRD 1543.[18]

It is shown there that an *efficiency index E* can be chosen which affords the possibility of rank-ordering different acoustic materials with respect to their ability to attenuate sound. The

efficiency index E is directly related to the MF and is given by the following equation.

$$E = \sqrt{\frac{RT^x}{\sigma_m}}, \qquad (4)$$

where R = flow resistance,
T = thickness of material,
x ··· ranges from 0.2 to 1.1 and is a function of the (lightweight) material,
σ_m = surface density of material alone.

To a good approximation, E is a characteristic constant for a given material for values of σ_m below about 0.15 lb per square feet.

It can be shown that, for sound treatment in aircraft, Structure V is usually somewhat superior to Structure III (see Figure 32). In Report OSRD 1543[15] information is supplied which aids in the selection of the most efficient structure of least surface density to achieve a given value of attenuation. After the structure has been chosen, procedures are outlined to find the minimum surface density σ_s which will yield the desired attenuation. These problems come up in connection with the design of sound treatment for an airplane which is required to reduce the levels in certain frequency bands below the maximum levels specified in the Army-Navy specifications. The matter of predicting the sound levels in the airplane before and after treatment and choice of structure is discussed in Section 2.2.3.

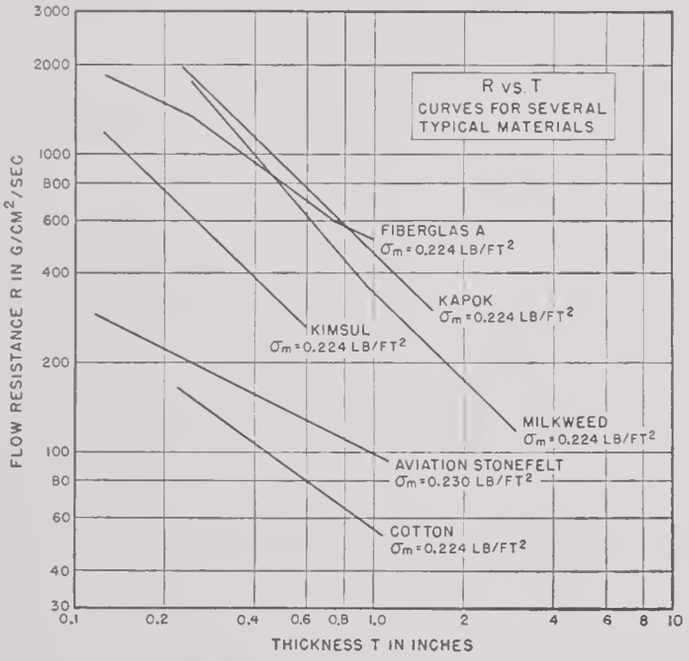


FIGURE 33. Measured flow resistance for various materials.

SOUND ABSORPTION

If a porous absorbent material is placed in a sound field, the sound pressure will tend to move the air particles contained in the pores of the material. Because of the frictional resistance between the air particles and the particles of the material some of the sound energy will be dissipated. It is evident that if this process is to take place efficiently, the material itself must be heavy enough to remain stationary, as a whole, or it must be mechanically constrained. The flow resistance of the material must be neither too small, which is an indication of insufficient frictional resistance, nor too large, in which case the particle motion would be too small. It can be shown that for a given surface density σ_m of the material there is an optimal value of flow resistance R for maximum sound absorption.

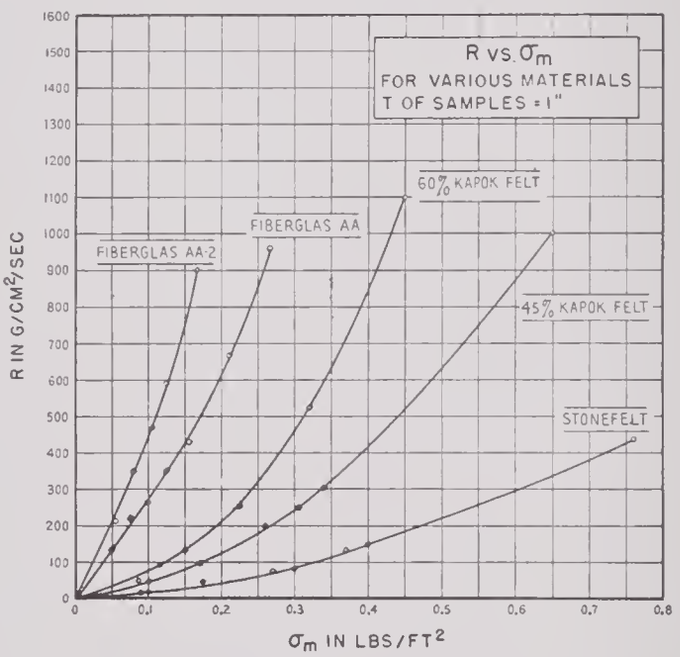


FIGURE 34. Measured flow resistance vs surface density for constant thickness.

Experiments have shown that the following conclusions can be drawn.

1. For efficient sound absorption at low frequencies, the spacing between the absorbed material and the Dural skin should be large.

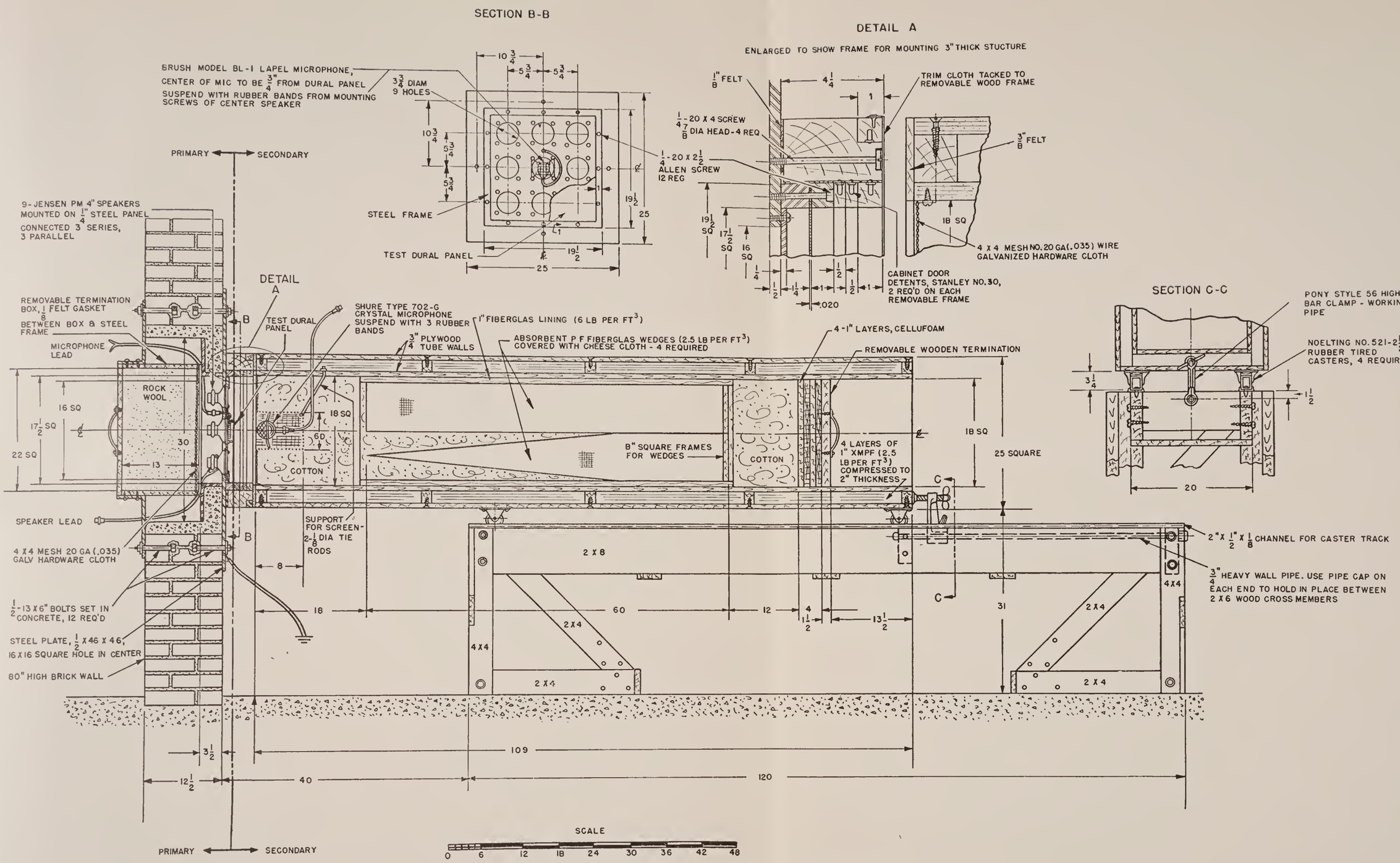


FIGURE 35. Apparatus for measuring acoustic attenuation.

2. The flow resistance should be less than 500 cgs units (g per sq cm per sec) for 1 in. of thickness.

3. The trim cloth covering the cabin side of the structure should have a flow resistance less than about 40 cgs units.

Flow Resistance

The flow resistance R of a porous material is defined as follows:

$$R = \lim_{v \to 0} \frac{p}{v} = \lim_{v \to 0} \frac{p}{V/At} \qquad (5)$$

in grams per square centimeter per second,

where p = steady pressure drop across the sample in dynes per sq cm,
V = volume (cu cm) of air passing the sample in t sec,
v = velocity of flow through sample,
A = area of sample in sq cm.

Since the flow resistance is a measure of the friction occurring between the fibers or particles of the material and the air passing through the channels between them, R depends upon the size, shape, and orientation of the channels and on the total surface area of the fibers in the material. These quantities depend, in turn, upon the diameter and cross-sectional shape of the fibers, the total number of fibers in the sample, and the manner in which the fibers are distributed and oriented in the material. One or more of these variables can usually be controlled in the process of manufacture of an acoustic material, and thereby R may be set at the required optimum value.

The general relation governing the steady flow of a compressible gas through a porous medium has been derived.[4] From this general theory one may derive a number of simple relations between the flow resistance R of a material and its physical characteristics, such as thickness, volume density, surface density, fiber diameter, and manner of construction ("felting").

The results of a large number of measurements of flow resistance carried out on a large number of lightweight acoustic materials show, however, that there are, in many cases, important deviations from the relation predicted by theory.

Figure 33 shows the empirically obtained relation between R and T for several materials. These relations can be approximated by the expression

$$RT^x = \text{constant}. \qquad (6)$$

The exponent x takes on a characteristic value for each material and surface density σ_m. The expression RT^x is, by equation (4), directly related to the efficiency index E for sound attenuation, discussed previously.

Figure 34 shows the empirical relation between flow resistance R and surface density σ_m, for a constant thickness T equal to 1 in. The curves were obtained by varying σ_m for a given material by packing more or less of it into samples of constant area. The variation of R under these conditions can be expressed as follows:

$$R\sigma_m^{-(1+x)} = \text{constant}, \qquad (7)$$

where x is the same as in equation (6).

For small values of σ_m (below about 0.15 lb per square foot) equation (7) can be approximated by

$$\frac{R}{\sigma_m} \approx \text{constant}. \qquad (8)$$

Equations (6) and (8) form the experimental basis for the validity of E in equation (4) as a fixed number for a given (lightweight) material.

2.2.3 Test Methods and Apparatus

Measurement of Sound Attenuation

In a report[32] recently published, a detailed description is given of the experimental evaluation of the sound-attenuating properties of structures designed for quieting aircraft. The method and apparatus described therein have proved useful and are, for the present purpose, preferable to the older methods described in the literature. In Section 2.2.2 the weight law [equation (2)] is arrived at by identifying the attenuation N in decibels with the ratio of the sound pressures in front and in back of the panel in decibels. Uniform pressure distribution on the "primary" side and a perfectly absorbing termination on the "secondary" side

were assumed. By extension, N is defined in the same way for more general structures where the weight law does not necessarily hold. The test equipment discussed in reference 32 is designed to measure the pressure ratio at the two sides of a structure under conditions approximating as closely as possible the assumptions made above. Residual discrepancies are removed by "calibrating" the equipment with a structure consisting of a single impervious septum (Dural 0.020 in. in thickness), for which the average attenuation N, at medium and high frequencies, is governed by the weight law.

The structure under test is clamped in a rigid steel frame of 17.5x17.5 in. inside dimensions and mounted on a steel plate fixed to a brick wall. A battery of nine loudspeakers mounted in a baffle and coupled to one *(primary)* side of the structure serves as a source of sound. The loudspeakers are backed by a box filled with rock wool which decreases sound transmission to the rear.

On the other *(secondary)* side of the structure under test there is located a long terminating tube. The tube is fitted with four long wedges of Fiberglas.[c] The space between the wedge is filled with loosely packed cotton. This termination not only excludes undesired sound pickup from the outside but approximates closely an ideal termination of perfect absorption.

To measure the primary and secondary sound pressures, one microphone is located on each of the two sides of the panel under test. Their electric output can be connected at will to a high-gain amplifier and filter. The paper drive of a graphic level recorder is mechanically coupled to the oscillator driving the loudspeaker through a suitable power amplifier. By means of the recorder the outputs of the primary and secondary microphones are recorded as a function of frequency.

Figure 35 shows a cross section through the apparatus. Sound source, panel under test, absorptive termination, and microphones are clearly shown. Figure 36 shows the battery of speakers and the various frames for assembling structures of 3 in. total thickness. The Dural septum is shown clamped in place in the lower half of Figure 36. For details of construction,

FIGURE 36. Primary and secondary sides of apparatus for measuring acoustic attenuation.

electric equipment, experimental technique, reference 32 should be consulted.

After the structure under test has been mounted in place and the absorptive termination sealed to the secondary side, the sound source is energized by a sinusoidal signal over the frequency range of from 100 to 10,000 c. By means of the level recorder, graphic records of the outputs of the two microphones are made in succession. Typical records using a logarith-

[c] A product of the Owens-Corning Fiberglas Corporation. For dimensions of the wedges see Chapter 10, Section 10.9.

mic frequency scale are shown in Figure 37 for a Dural panel of 0.020-in. thickness. By point-by-point subtraction, taking due account of amplifier gains and microphone corrections, a curve of attenuation N versus frequency

suitable averaging method, from experimental data obtained as described previously. By planimetering the area under the two experimental curves obtained from the primary and secondary microphone outputs, the average at-

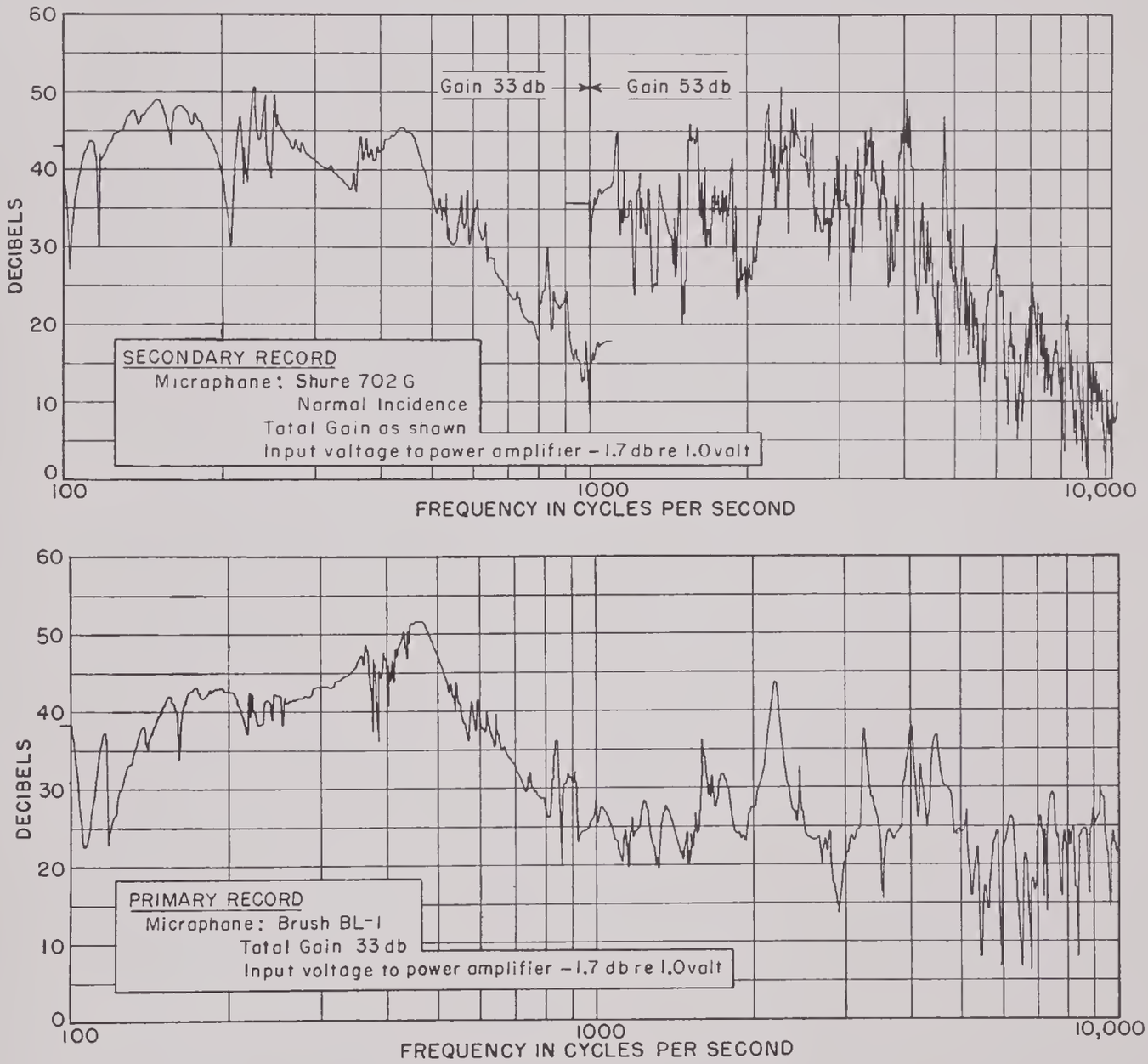


FIGURE 37. Records of the sound pressures on the primary and secondary sides of the structure under test.

(similar to Figure 30) can be obtained. Clearly, an averaging method for evaluating graphs of such fluctuating character is required.

To describe the attenuating properties of a structure in practical terms for the purposes at hand, it has been found useful to obtain the *average* attenuations N_3 and N_5 at frequencies 3,000 and 5,000 c, respectively, by a

tenuation values in a given frequency band can be obtained by subtraction of the two areas. The following octave bands have been successfully used for this averaging procedure: 1,500 to 3,000 and 3,000 to 6,000 c for N_3; 2,500 to 5,000 and 5,000 to 10,000 c for N_5.

By entering the surface density σ of the total structure (including Dural skin) into the

family of graphs for weight law given in Figure 29, the attenuation values at 3,000 and 5,000 c predicted by the weight law can be obtained. The expressions n_3 and n_5, the difference in attenuation at 3,000 and 5,000 c between the average measured attenuations and weight law, are a measure of the attenuation properties of the structure. In Figure 38, N_3, N_5, n_3, and n_5

Test No.	Material	Mfr.	Type of Struc.	MATERIAL Thick. Inches	MATERIAL Weight Pounds	Lbs/ft²	Septum Weight Lbs/ft²	Trim Cloth Weight Lbs/ft²	Treatment Wt. Lbs/ft²	Total Struc. Weight Lbs/ft²	Total Attenuation 3000 cps N_3	Total Attenuation 5000 cps N_5	DB Better Than Weight Law n_3	DB Better Than Weight Law n_5
1	3" AIR SPACE		For Cal.				0	0	0	.273	35.5	40.0	0	0
2	3" AIR SPACE, TRIM						0	.057	.057	.330	35.9	40.9	-1.5	-0.8
3	3" AIR SPACE, SEPTUM						.061	0	.061	.334	42.7	49.2	5.5	7.5
4	2" AIR SPACE, SEPTUM						.061	0	.061	.334	43.6	49.8	6.4	8.1
5	2" AIR SPACE, SEPTUM, 1" AIR SPACE; TRIM		III	1" air space			.061	.057	.118	.391	43.6	50.1	5.0	7.0
6	AA	O.C.	III	1/2	.052	.024	.061	.057	.142	.415	43.2	53.1	9.1	14.5
7	AA	"	III	1	.104	.049	.061	.057	.167	.440	52.3	65.0	13.2	20.9
8	AA	"	V	1	.104	.049	.061	.057	.167	.440	54.9	63.0	15.3	23.9
9	XMFF	"	III	1/2	.187	.091	.061	.057	.209	.482	46.4	54.7	6.0	9.8
10	XMFF	"	III	1	.385	.187	.061	.057	.305	.578	50.3	60.5	8.3	14.0
11	XMFF	"	V	1	.385	.187	.061	.057	.305	.578	54.2	67.1	12.2	20.6
12	XMFF	"	III	2	.761	.353	.061	.057	.476	.749	52.6	64.7	8.3	15.9
13	XMFF	"	V	2	.761	.353	.061	.057	.476	.749	62.8	75.1	18.5	26.5
14	RAYON B	DuP.	II	5/8	.331	.179	0	.057	.236	.509	41.4	47.7	0.5	2.3
15	RAYON B	"	III	5/8	.331	.179	.061	.057	.297	.570	46.9	56.4	5.0	10.0
16	RAYON B	"	II	1-1/4	.732	.363	0	.057	.425	.693	43.9	56.3	5.2	8.1
17	RAYON B	"	III	1-1/4	.732	.363	.061	.057	.436	.759	52.1	61.8	7.7	12.9
18	RAYON B	"	V	1-1/4	.732	.363	.061	.057	.436	.759	60.1	72.4	15.7	23.5
19	KAPOK NO. 1	P.C.	III	1-3/4	.706[a]	.332[a]	e	.057	.389	.662	53.4	70.0	15.2	22.3
20	KAPOK NO. 2	"	III	1-1/4	.626[b]	.294[b]	b	.057	.351	.624	56.6	67.1	13.9	19.9
21	KAPOK NO. 2	"	III	1-1/4	.626[b]	.294[b]	.061	.057	.412[c]	.685	58.2	72.8[d]	14.7	25.5[d]
22	KAPOK K3		V	2	.880	.414	.061	.057	.532	.805	75.3	95.9[d]	30.4	46.5[d]
23	COTTON FELT NO. 1		III[e]	1/2	.282	.133	.061	0	.194	.467	47.6	53.4	7.4	13.7
24	COTTON FELT NO. 3		III[e]	1/2	.377	.177	.061	0	.238	.511	48.0	53.9	7.1	13.5
25	M STONEFELT	J.M.	III	1	.534	.251	.041[f]	.057	.349	.622	49.2	58.6	6.5	11.4
26	M STONEFELT	"	III	1	.534	.251	.061	.057	.369	.642	49.9	59.4	7.0	12.0
27	5DFC	O.C.	V	1	.179	.084	.061	.057	.202	.475	48.9	59.9	8.6	15.1
28	6DFC	"	V	1	.290	.136	.061	.057	.254	.527	51.7	64.6	10.5	18.9
29	7DFC	"	V	1	.376	.177	.061	.057	.295	.563	53.6	66.6	11.7	20.2
30	3DFG-1	"	V	2	.347	.163	.061	.057	.231	.554	55.3	66.1	14.1	19.9
31	D-4	"	V		.172	.081	.061	.057	.199	.472	48.7	58.3	8.4	14.0
32	G-4	"	V	1	.278	.131	.061	.057	.249	.522	50.6	60.7	9.4	15.0
33	WOOD CONV.	W.C.	III	1	.412	.194	.061	.057	.312	.585	47.3	56.0	5.2	9.4
34	STRATOSEAL	E.I.	III	1	.763[f]	.359[f]	f	.057	.416	.689	64.4	73.5	20.9	30.5
35	FIBERGLAS B	O.C.	III	1	.415	.195	.061	.057	.313	.536	57.6	70.6	15.5	24.0

NOTES: All tests made with an .020" dural plate with a surface density of .273 lbs/ft².

Asbestos Septum with surface density of .061 lbs/ft² was used for all tests except Test Nos. 1, 2, 14, 16, 19, 20, 23, 34.

Aristo trim cloth with a surface density of .057 lbs/ft² was used for all tests except Test Nos. 1, 3, 4, 23, 24.

Depth of structure 3" for all tests.

a. Weight includes attached paper septum.
b. Weight includes attached asbestos septum.
c. Structure had attached septum plus a second adjacent septum.
d. Attenuation found by extrapolating 6000-10,000 cps portion of 2500-10,000 cps band.
e. No trim cloth used.
f. Magicskin septum.

ABBREVIATIONS USED FOR MANUFACTURER'S DESIGNATION

O.C. - Owens Corning Fiberglas Corporation
W.C. - Wood Conversion Company
E.I. - Elavaty Insulation
DuP. - E. I. duPont de Nemours and Company
P.C. - Philip Carey Manufacturing Company
J.M. - Johns Manville

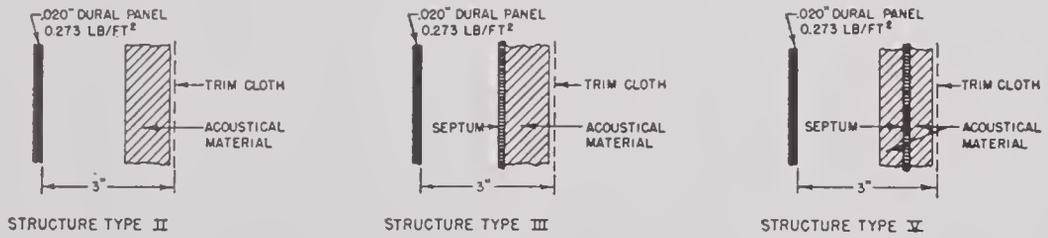


FIGURE 38. Numerical values of attenuation for various materials and structures.

are tabulated for a number of materials and structures.

Figure 39 shows plots of N_3 versus N_5 and n_3 versus n_5 for the materials and structures tabulated in Figure 38.

Figure 40 shows measured average values of N for two single impervious septa, in various

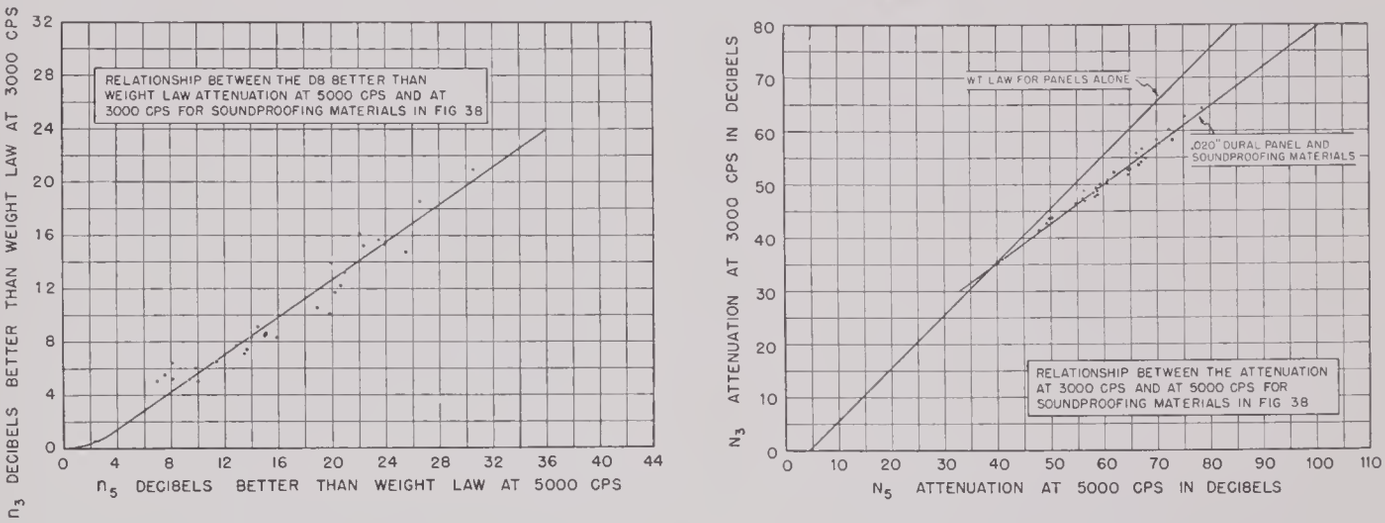


FIGURE 39. Relation between N and n for various materials and structures.

frequency bands for certain positions of the primary and secondary microphones. The measured values fall below the values predicted by theory (weight law). The experimental apparatus was "calibrated" by applying the corrections from Figure 40 for the case of the 0.020-in. Dural panel to adjust it to weight law.

MEASUREMENT OF SOUND ABSORPTION

The sound-absorbing properties of a structure have already been generally discussed in terms of an absorption coefficient α (see Section 2.2.1). The absorption coefficient α is, among other things, a function of the angle of incidence of the sound waves and has physical meaning as originally defined only in random sound fields.[5, 19] The absorptive properties can also be described in terms of the *acoustic impedance* of the structure. Although the acoustic impedance of absorbent structures can be determined experimentally,[3] preference has been given here to the absorption coefficient.

Two methods of measuring α are commonly in use. The first is the method approved by the Acoustical Materials Association[7] and is especially applicable to ordinary room acoustics. Large samples of the structure to be tested are mounted in a reverberation chamber. After exciting the chamber with a sound source, the reverberation time is measured after the source has been turned off. From the reverberation time, the room volume, and the surface area of the samples, α can be computed. The absorption coefficient so determined is physically meaningful in the light of the original definition, since it was determined under more or less random conditions of wave motion in the room.

It was deemed advisable, in addition, to measure the absorptive properties of structures in terms of a quantity obtained from measurements in a tube. A sample of the structure (8x8 in.) is mounted at one end of a square steel tube.[18] A loudspeaker is mounted at the other end. When the sound source is energized by a sinusoidal voltage, a standing wave pattern is set up in the tube which can be explored by a movable microphone. From the ratio of the maximum and minimum microphone voltages, as the microphone is moved along the tube axis, a measure of the absorption coefficient α for normal incidence can be obtained. The frequency range is limited, since the simple theory assumes, among other things, uniform pressure distribution in any cross section of the tube.

Reference 18 contains a number of graphs of α vs frequency in the range from 100 to 2,000 c obtained by the tube method and also a few data from chamber measurements. The data show that, at frequencies of 1,000 c or higher, the absorption coefficient is at least

0.5 or greater for most materials. The contours of Figure 38 show that the sound-level reduction is not likely to be improved significantly by even large errors in α, provided α is greater than 0.5. The values of α assigned to structures for computational purposes are a "weighted average" of the values obtained by the two methods (see Section 2.3).

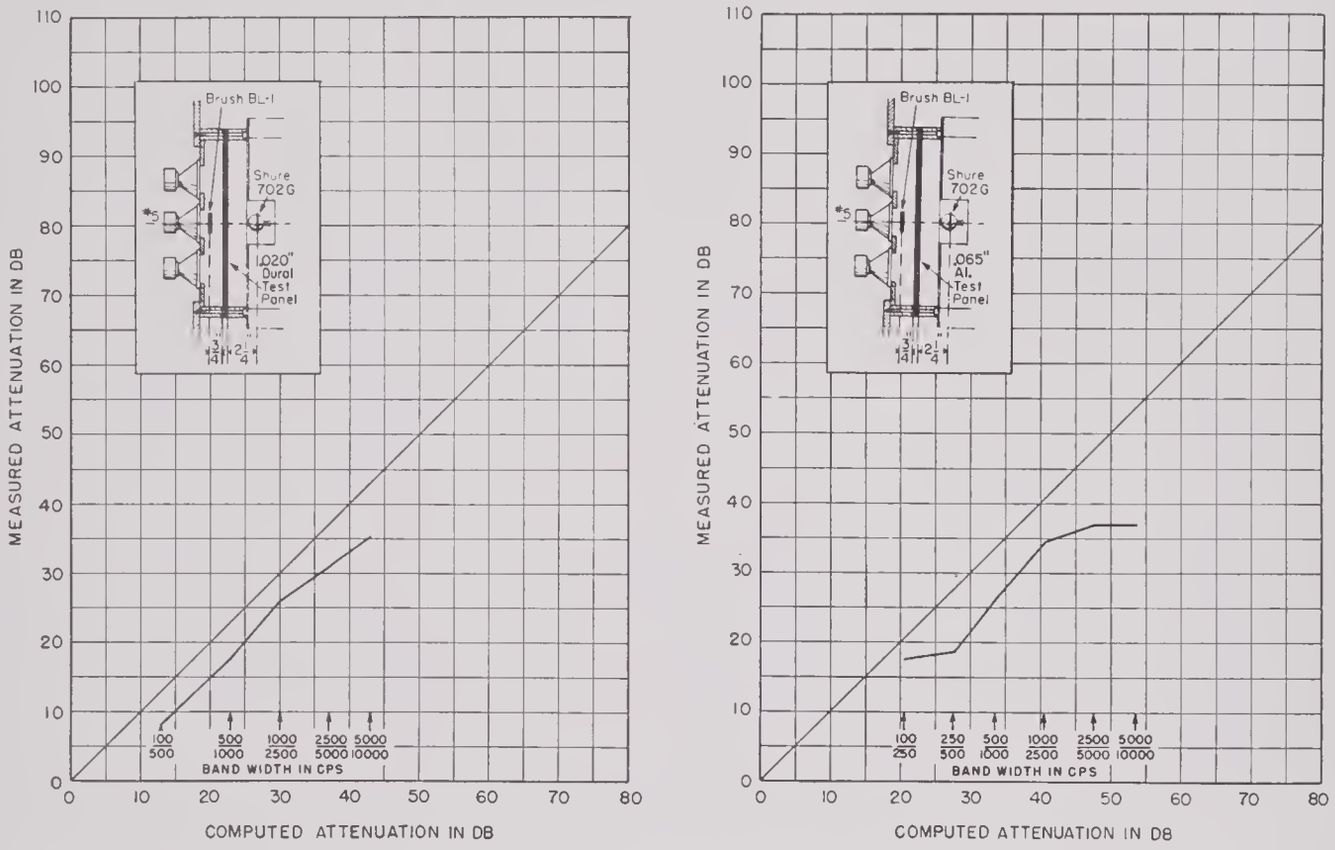


FIGURE 40. Measured vs computed attenuation N for two panels (0.020 in. and 0.065 in. thickness).

MEASUREMENT OF FLOW RESISTANCE

The apparatus used for the determination of the flow resistance R consists of the following components.

1. A holder for the sample.
2. A means of drawing air through the sample at a uniform rate.
3. A gauge for measuring the pressure drop across the sample.
4. A gauge for measuring the rate of volume flow of air through the sample.

Figure 41 shows a picture of some of the components listed above. Circular samples of $2\frac{5}{32}$ in. in diameter are used. Details of the procedure and technique can be found in Report OSRD 1543.[18]

To obtain the efficiency index $E = \sqrt{RT^x/\sigma_m}$ it is necessary to determine x by measuring R for various thicknesses T. The efficiency index E is usually taken for $\sigma_m = 0.1$ lb per sq ft.

Army-Navy Specifications AN-S-32_a contain instructions for a battery of further tests of materials to test conformity with the general requirements for acoustic materials and structures in Section 2.2.2.

2.3 PREDICTIONS OF AMBIENT NOISE LEVELS IN AIRCRAFT

From analysis of numerous data on ambient noise levels and spectra in multi-engine military aircraft accumulated during the years of 1940 through 1944, an empirical method of predicting noise levels has been developed at the Electro-Acoustic Laboratory. A set of empirical graphs has been derived taking into

account important variables such as engine power, propeller-tip speed, and amount and quality of acoustical treatment. It should be borne in mind that, through limitations of the original data from which the contours have

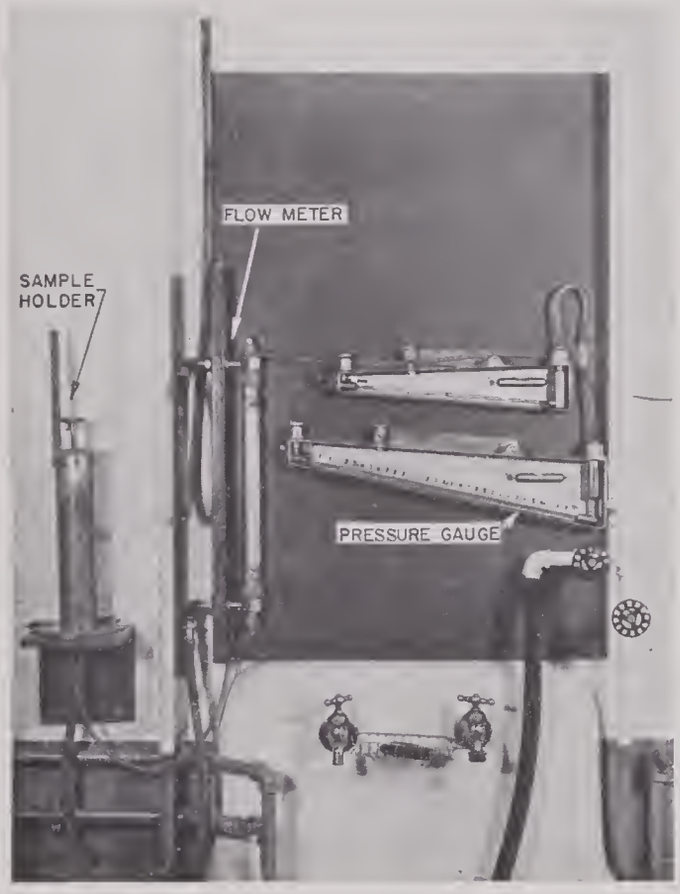


FIGURE 41. Experimental arrangement for measuring the flow resistance of acoustic materials.

been obtained, these contours apply primarily to the multi-engine aircraft, *now obsolete*, which were in use in the closing years of World War II. They can be used as a rough guide only for aircraft of more modern design. In addition, the following assumptions were made:

1. Two- or four-engine aircraft.
2. Normal cruising power conditions.
3. Three-blade propellers.
4. Collector ring exhausts.
5. Perfect seals around windows, hatches, etc. No secondary noise sources present in the cabin.
6. Surface density of sound treatment (if any) less than 0.5 lb per sq ft.
7. The reference point at which levels are computed is within 6 ft of the propeller plane, about 2 ft from the fuselage at the head level of a sitting man.

In Figure 42 are shown typical spectra in a bomber, treated and untreated. For purposes of calculation, the frequency range has been divided in three regions, *A*, *B*, and *C*, as indicated.

The level in region *A* can be predicted on the basis of engine power, propeller-tip velocity, minimum propeller-tip separation from the fuselage, and airspeed. Sound treatment of surface density less than 0.5 lb per sq ft does

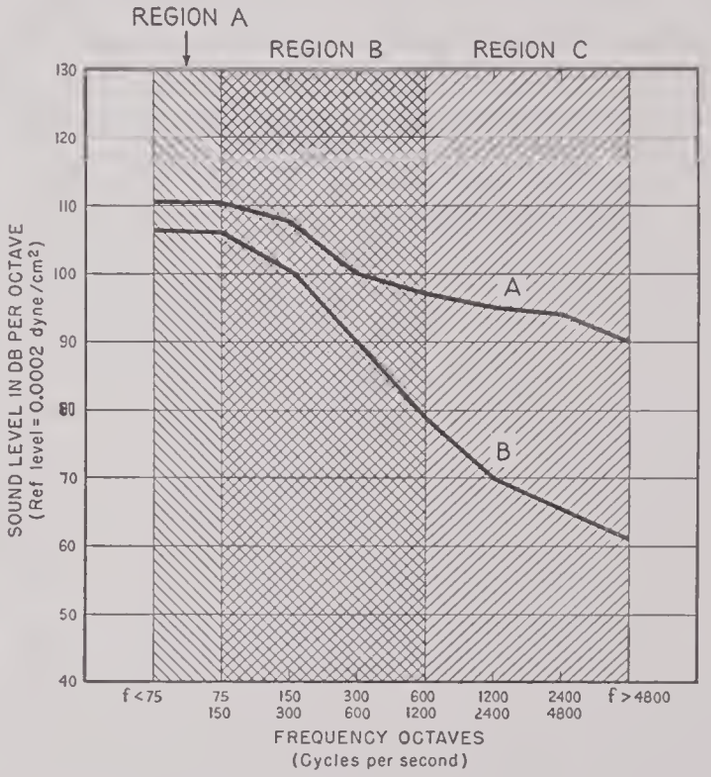


FIGURE 42. Typical noise spectra in untreated (A) and treated (B) bomber.

not reduce the levels in this region appreciably. Empirical graphs based on experimental data are given in Figures 4 and 10 of Report OSRD 1543[18] for purposes of computation of sound levels.

In regions *B* and *C* calculations consist in arriving at the slope of the spectrum from a computation of the ratio of total noise-absorbing area *A* to total noise-transmitting area *T* of the cabin. The total absorption *A* is simply the sum of the individual areas multiplied by the appropriate absorption coefficient α. The procedure outlined in Report OSRD 1543[18] requires computation at 1,000 and 3,000 c. As

an average value of α for these frequencies the values 0.8 and 0.9 can be assumed. The total transmission T is the sum of the individual areas multiplied by the appropriate transmission coefficient τ. The transmission coefficient τ can be arrived at for 1,000 and 3,000 c from N_1 and N_3, taken from Figure 38 with the help of equation (1). Figures 7 and 8 in Report OSRD 1543[18] are empirical contours giving two values of spectrum slope for different values of A/T at 1,000 and 3,000 c. As a result, the spectrum to be expected is bracketed by two computed spectra. Figure 43 shows measured and computed spectra for four different aircraft and gives an estimate of the agreement of the predicted spectra with the computed spectra which can be expected. One of the reasons for the large discrepancy in the case of the A-20-B airplane is its use of ejector exhausts.

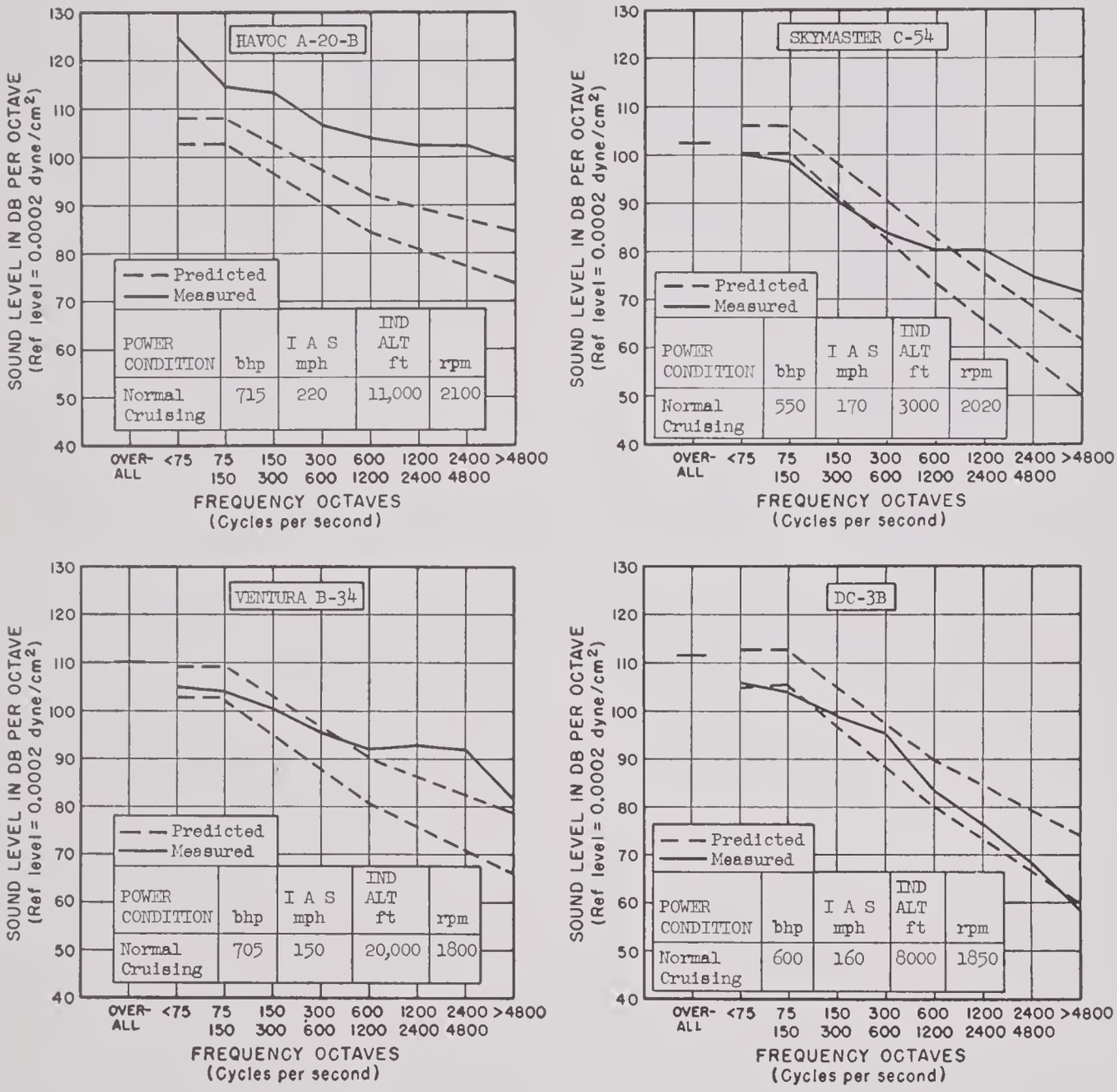


FIGURE 43. Measured and predicted noise spectra in airplanes.

The computational procedure outlined above is useful as a guide either in the case of untreated aircraft or in the case where the treatment is known and specified. Army-Navy Speci-

fications AN-S-33$_a$ require that the sound level in the frequency band 1,200 to 2,400 c be less than 80 db at the head position of the crew members. If this requirement is to be attained in a given airplane, it becomes necessary to attempt the design of the *optimum* sound treatment, since weight is at a premium. The minimum surface density σ and the minimum efficiency index E necessary to meet the specifications for a given airplane must be found. Procedures for accomplishing this are contained in Part 3 of Report OSRD 1543.[18] By successive approximations a treatment of minimum total added weight can be arrived at.

2.4 RECOMMENDATIONS FOR NOISE REDUCTION AND CONCLUSIONS

The discussion in the preceding sections will now be made the basis for a brief list of recommendations for installing optimum sound treatment in certain combat vehicles. Examples have been chosen in accordance with their general importance. At the same time, their choice is conditioned by the assignment of particular problems to the Electro-Acoustic Laboratory for solution. Judicious application is possible and desirable to other cases not listed.

2.4.1 Aircraft

1. Noise reduction of the medium- and high-frequency components of the noise is necessary to avoid impairment of voice communications and reduce annoyance.

2. The effectiveness of sound treatment in an airplane cabin is determined primarily by the *attenuation* of transmitted sound of the structure, the external sound sources, and the *absorption* of sound by the structure.

3. The effectiveness of a sound treatment is, at best, only as good as the weakest parts of the cabin walls transmitting sound from the outside. Hence windows, floor, and untreated portions of the Dural skin must be sealed and sufficiently heavy so that the sound transmitted by these areas will not contribute significantly to the noise level in the cabin.

4. For minimum noise levels, blowers, auxiliary power plants, and other internal noise sources must be effectively silenced.

5. In multi-engine airplanes of the type used in the closing years of World War II, it is possible to reduce the noise levels to meet Army-Navy Specifications AN-S-33$_a$, if sufficient weight of treatment is added.

6. For a given airplane, there is a minimum surface density of treatment to meet the specifications if a material with an efficiency index greater than a minimum value is used.

7. The minimum weight of added treatment can only be realized if points (3) and (4) are observed.

8. Optimum structures simple to install and to fabricate are given in the Army-Navy Specifications AN-S-33$_a$.

9. Requirements for installation and mounting of the treatment are contained in Army-Navy Specifications AN-S-32$_a$.

10. The acoustical requirements should be made part of the basic design considerations of an airplane.

2.4.2 Combat Information Centers Aboard Ship

1. Noise reduction of the medium- and high-frequency components of the noise is necessary to avoid impairment of voice communication.

2. The effectiveness of sound treatment in a closed compartment on shipboard depends primarily on the absorption of sound by the structure. Owing to the presence of heavy bulkheads and decks, sound transmission is usually inconsequential.

3. Internal noise sources: blowers, typewriters,[26] etc., must be effectively silenced.

4. A blanket of certain absorbing materials,[23] 2 in. thick, spaced 1 in. from the bulkhead or ceiling, is recommended.

5. The acoustic material should have a flow resistance between 20 and 30 g per sq cm per sec per in. thickness.

6. Open metal mesh covers with a large ratio of area of holes to total area or trim cloths (R less than 20 g per sq cm per sec) are suitable to cover the material.

2.4.3 Fortifications

1. The reduction of the frequencies of shock and explosion noises, which comprise mainly the low end of the audible frequency band, is desirable to avoid detrimental effects on combat personnel[27] and voice communications.

2. To provide a structure with high absorption coefficient down to low frequencies, the thickness must be large. Four inches is recommended as the minimum thickness.

3. As large an area as possible should be treated.

4. Ventilation ducts, gun embrasures, etc., should be lined with acoustic material, and acoustic leaks to the outside should be minimized.

5. Aural protective devices should be worn by the personnel (see Section 2.6).

6. In addition to acoustical treatment, insulation against mechanical shock is desirable.

2.5 THE EFFECT OF NOISE ON PSYCHOMOTOR EFFICIENCY

One of the first military problems undertaken by the Psycho-Acoustic Laboratory was to discover the effects of intense noises on psychomotor efficiency. It was believed that by measuring the various aspects of a subject's mental, motor, and physiological behavior under the stress of noise and vibration, one should be able to determine the practical degree to which this environmental stress should be eliminated from combat situations.

Apparent from the beginning was the extreme difficulty of evaluating in a quantitative manner the role of noise in the great complex of variables and conditions affecting all human performance. Acceptable standardized tests of even so obvious a phenomenon as human fatigue were largely nonexistent, and many efforts to probe the psychophysiological effects of internal and external stresses have been defeated by the overwhelming complexity of the factors conditioning human activities. It was decided, therefore, that the first effort should go toward the design and development of psychological and motor tests capable of sampling and evaluating a wide range of tasks which might be required under stress of noise.

The noise stress which was used for most of the tests simulated the noise encountered in the cabin of a typical bombing plane. A voltage of uniform spectrum can be produced by ionization in a gaseous tube, and a low-frequency voltage rich in harmonics can be generated by a relaxation oscillator. When these two voltages are properly mixed, amplified, filtered, and equalized, and led to a battery of loudspeakers, the resulting sound has a spectrum similar to that of an airplane noise.

The spectra of the two simulated airplane noises used in the following experiments are

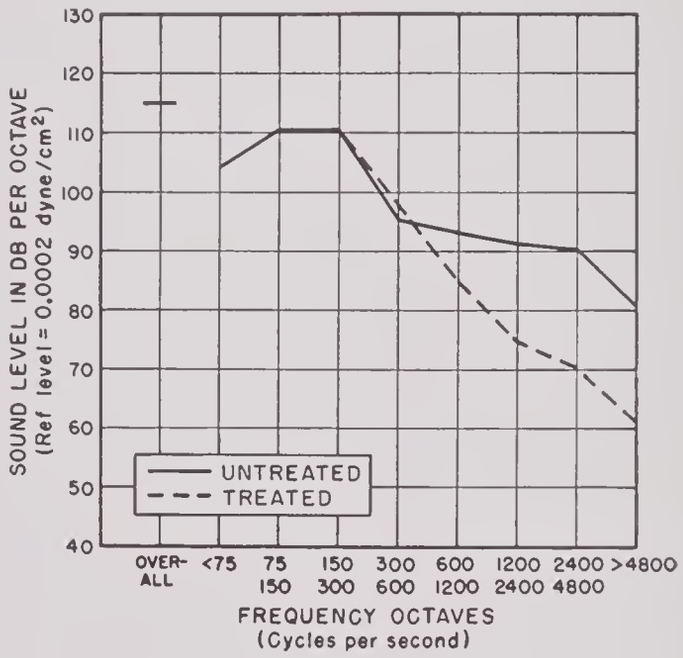


FIGURE 44. Sound level per octave band for the two synthetic airplane noises used in the tests of psychomotor efficiency. The spectra were designed to simulate the noise in acoustically untreated and acoustically treated planes. See Figure 42.

shown in Figure 44. These spectra were designed to simulate the typical noise conditions in acoustically untreated and acoustically treated planes. (See Figure 42.)

The battery of psychomotor, physiological, and psychological tests included measurements of the near point of vision, speed of accommodation, speed of eye movement, visual acuity, muscular tension, heart rate, finger tremor, blood pressure, marksmanship, card sorting, span of apprehension, paper form board tests, steadiness tests, tapping board and standard

pursuit rotors for testing motor coordination, coordinated serial pursuit meters which partially simulated the requirements of instrument flying, tests of coordinated serial reaction time, audiometric measurements of hearing loss, vibration tests, tests of serial disjunctive reaction time, coding tests, judgments of distance, etc. With this extensive battery it was possible to take a very thorough sample of the subject's behavior both in the presence of noise and in the quiet.

The results of a preliminary series of tests in which subjects were exposed for three hours to a noise (approximately 115 db) demonstrated that acoustic stress is detrimental to certain aspects of performance but leaves other aspects unimpaired.[6] Since the effects of the noise on psychomotor efficiency were in most cases very small, and because there was some evidence for a greater loss in proficiency at the end than at the beginning of the three-hour experiments, it was necessary to test the implication that longer exposure to the noise would exhibit these effects still more conspicuously. Consequently, a second series of experiments was designed in which the effects upon psychomotor efficiency of continuous seven-hour periods of work in an airplane noise of 115 db could be evaluated by comparison with the effects of similar periods of work in an airplane noise of only 90 db.[9] The intensity of 90 db is just about sufficient to mask the casual noises of the laboratory and to discourage conversation among the subjects.

Five carefully selected subjects were employed for a period of two months in these experiments. During the first few weeks, the subjects were tested in miscellaneous ways and allowed to practice the performance tests in order to reach a plateau in learning from which later improvement due to practice would be minimized. Following this preliminary period, the subjects were tested in the main series of experiments requiring four consecutive days in each of four consecutive weeks. In this series, the conditions of "noise" (115 db) and "quiet" (90 db) were distributed over the sixteen experimental days in such a way as best to counterbalance such systematic variables as learning, boredom, etc. On the other days of the week, tests involving procedures not compatible with this counterbalanced order were applied. Most of the tests were so intensive, the control so adequate, and the counterbalanced order so effective that differences in performance of the order of 1 per cent could be detected with high reliability.

In tests of this nature, the importance of maintaining a stable level of motivation for the subject cannot be overemphasized. Perhaps the influence on efficiency of this factor of motivation may best be demonstrated by an example from the results during the practice period for the coordinated serial reaction-time test. On one of the practice days, the experimenter noticed that the scores during the eighth period were much better than for the preceding periods of that day. The subject was asked later if he could account for this fact. He reported that by self-instruction, his general level of muscular tension had been greatly increased. The experimenter could see that the subject's grip on the chair with his free hand was tightened, and his facial expression revealed considerable muscular tension. The subject reported that he could not possibly maintain such a level of activity for an extended period, but that he had tried for the one block of trials to push himself to extreme effort.

The subject was, therefore, asked each day for four consecutive days at the beginning of the eighth period to duplicate as well as he could the exact conditions of tension and effort experienced on the first day, and to run through the sequence as rapidly as possible. As the curve of Figure 45 shows, this increased motivation resulted in a striking improvement in the average performance of the eighth period. The average for the succeeding blocks, however, returns to the subject's typical level of performance.

Since there is no way to control completely all fluctuations in effort, any experiment in which one hopes to minimize the effect of changing motivational conditions must rely on a large number of scores taken under the same conditions. If the fluctuations are not systematic with respect to changes in noise level they can be expected to cancel out, but only if a large number of scores are taken.

The results of these experiments agree with common sense. Although, as a *subjective experience*, noise is disagreeable and tiring, most types of mental, motor, and physiological activities are affected very little by noise as such. Just as when a man steps from an airplane after a long flight at moderate altitude, he is

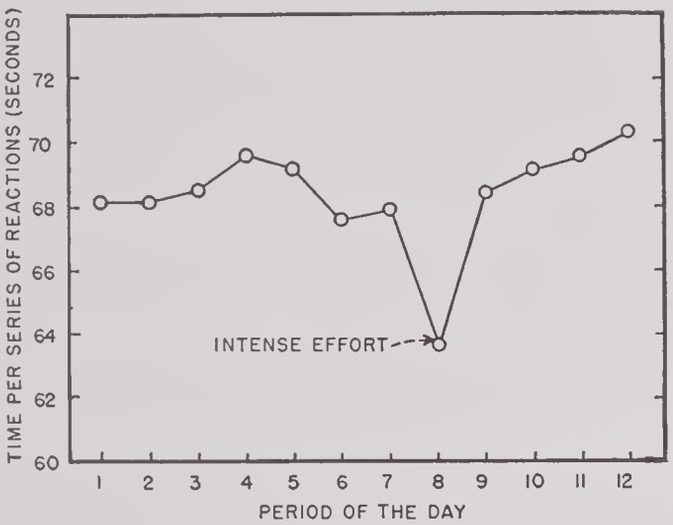


FIGURE 45. Showing the change in performance introduced by intense effort during one of a series of 12 trials.

able to walk, talk, write, think, etc., in about the same way as before the take-off, so in the laboratory no dramatic change in behavior follows exposure to airplane noise. Confirming evidence for this general conclusion was obtained at Stevens Institute of Technology in an investigation of sound as a military weapon.[13] It is found, in general, that communication is difficult or impossible, hearing losses result, and subjective experiences of fatigue and annoyance are reported, but no conclusive evidence appeared to confirm the notion that intense noise interferes with the performance of motor tasks.

2.5.1 Tests Showing Detrimental Effects of Noise

Temporary Hearing Loss

Long exposure to airplane noise, similar to that encountered in a bombing plane having little or no acoustical treatment, results in decreased auditory sensitivity (stimulation deafness). These hearing losses are greatest in the region from 3,000 to 5,000 c, but after eight hours of continuous exposure, losses as great as 30 db occur at frequencies as low as 250 c. In all cases studied these impairments proved temporary, but the fact of their occurrence is proof of the stress imposed on the auditory organ. As the duration of the exposure is increased, the stress, as manifested in hearing loss, is increased and the time required for recovery is prolonged.

Stimulation deafness is related to three obvious aspects of exposure to complex noises. These are (1) the overall intensity of noise, (2) the shape of the noise spectrum, and (3) the duration of exposure to noise.

The amount of change in the threshold is directly related to the intensity of the noise. The most severe impairment results at the higher frequencies. This becomes even more striking upon consideration of the fact that the noise spectra used for these exposures have by far the greater part of their energy confined to the lower frequencies. The low frequencies are not discernibly impaired as a result of stimulation for 30 minutes by intensities below 120 db.

The effect of the shape of the spectrum on the hearing loss is illustrated in Figure 46. In

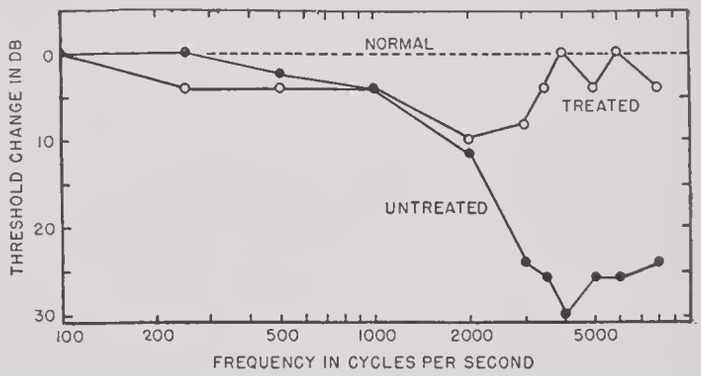


FIGURE 46. Illustrating hearing losses resulting from 30 min exposure to 110 db of two synthetic airplane noises.

this figure, the noise spectrum used to obtain the hearing loss indicated by the closed circle was similar to that encountered in a bombing plane without acoustical treatment. The hearing loss indicated by the open circle was obtained using a spectrum of noise similar to that

encountered in a bombing plane with acoustical treatment (see Figure 44). Acoustical treatment attenuates the high-frequency components of the noise, and the effect of this difference in high-frequency energy is apparent from the two curves.

The effects of the duration of exposure to the noise are shown in Figure 47 and Figure 48. Figure 47 shows the effect of short exposure

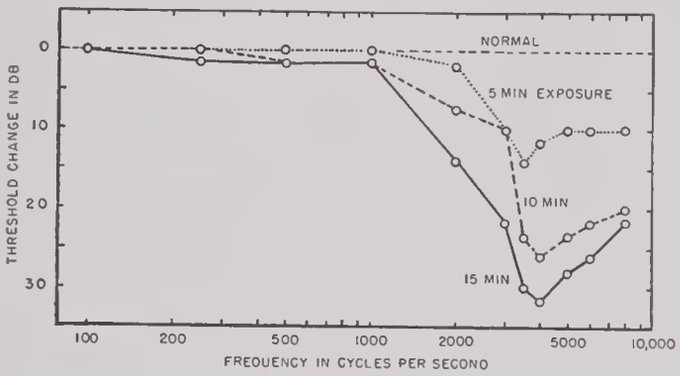


FIGURE 47. Showing the relation between the severity of hearing loss and the duration of exposure to 105 db of white noise.

to a noise having uniform distribution of energy throughout the frequency range and Figure 48 shows the effect of long duration of exposure to a spectrum similar to that en-

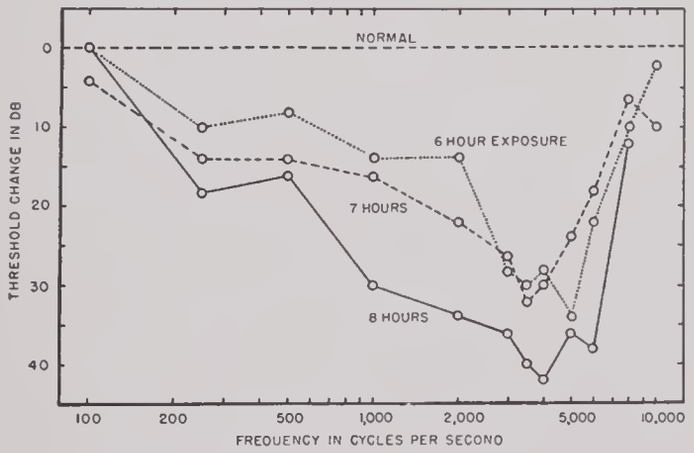


FIGURE 48. Illustrating hearing loss caused by long exposures to 115 db of noise simulating a bomber without acoustic treatment. With long exposures the effect spreads to the lower frequencies.

countered in an acoustically untreated bomber. In both cases it can be seen that the amount of hearing loss increases as the length of exposure is increased, and that with the longest exposures, the hearing loss spreads to the lower frequencies.

Audiograms taken during the course of recovery from exposure to airplane noise one-half hour at 110 db are shown in Figure 49. In

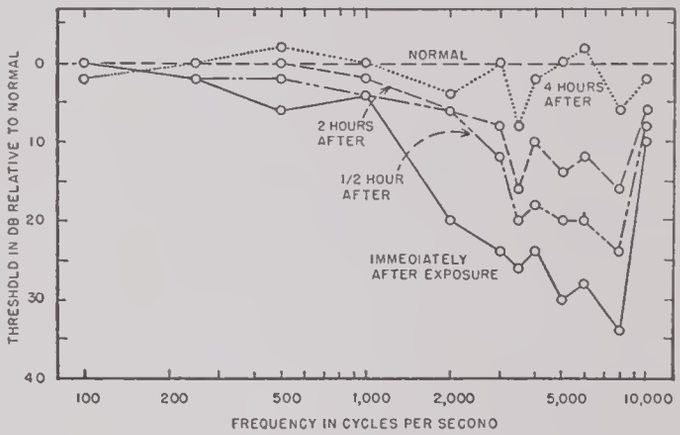


FIGURE 49. The course of recovery from one-half hour exposure to 110 db of noise from a plane without acoustic treatment.

general, the recovery is most rapid immediately after exposure. The higher frequencies suffer greater damage and recover more slowly than the low frequencies.

SUBJECTIVE EXPERIENCES ASSOCIATED WITH NOISE

The subjects were required to fill out questionnaires daily regarding their impressions. Uniformly, they expressed decided preference for the days on which they worked in quiet (90 db). They showed a tendency to report greater feeling of fatigue at the end of the experiments in noise, but this report was not consistent. After every experiment in noise, all subjects had reported ringing in the ears (tinnitus), but in only one instance did a subject report tinnitus on a quiet day.

The subjects also reported that some noises were more annoying than others. For example, it was found that the noise from a bomber having acoustical treatment was much less annoying than the noise from an untreated plane. In order to make these two noises equivalent in annoyance value, it is necessary to make the noise of the treated bomber 10 db more intense than the noise from the untreated bomber.

In order to investigate the factors underlying these differences in annoyance, an analysis was made of the relative annoyance produced by various frequency bands of noise.[20] A "white" or uniform noise containing all frequencies from 40 to 9,000 c at approximately equal intensity was divided by an appropriate filter system into twelve bands. The frequency band from 1,900 to 2,450 c was used as the standard noise band and subjects were required to equate each of the other eleven bands to this standard band with respect to (1) loudness and (2) annoyance. The results of this procedure are shown in Figure 50. In this figure, loudness is

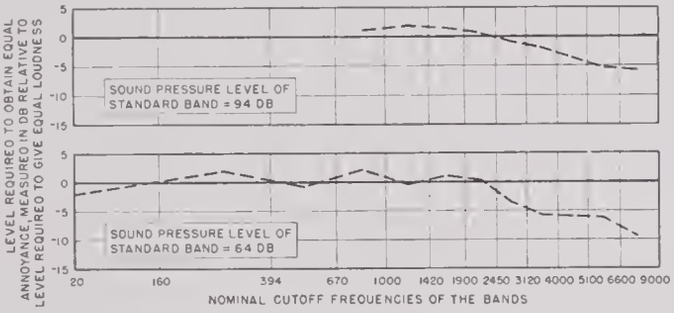


FIGURE 50. Difference between loudness and annoyance judgments. Loudness is plotted as a straight horizontal line for both intensity levels. The dashed line is drawn through the average difference in decibels between the loudness and annoyance judgments for five subjects.

plotted as a straight line for each of four intensity levels at which judgments were made, and the difference in decibels between the loudness and annoyance judgments for each of five subjects is shown by separate points for each band. In general, the results show that loudness and annoyance are separate and distinct discriminable characteristics of sound. Among bands adjusted to be equal in loudness, those bands of noise composed of high frequencies are definitely more annoying than the standard band.

The relatively greater annoyance of the high frequencies is also true for tonal signals.[28] In general, studies of annoyance have shown it to be directly related to three important variables, (1) the loudness of the noise, (2) its pitch, and (3) modulation of the pitch and loudness. The most annoying sounds are those whose energy is concentrated in the high frequencies and whose pitch or loudness is changing randomly and unpredictably.

Intelligibility of Speech

The most serious hazard of airplane noise arises from its interference with communications. Intense ambient noises in airplanes interfere not only with direct person-to-person communication but also are picked up in microphones and leak in around earphone cushions. Because of its importance, the investigation of noise interference with communication systems became a major research problem, and the results are treated in considerable detail in the following chapters.

2.5.2 Tests Showing Indeterminate Results

Certain tests gave inconclusive results, owing to large individual differences among the subjects, to wide variability among the results from a single subject, or to certain difficulties in the experimental procedures.

Coordinated Serial Reaction-Time

The coordinated serial reaction-time apparatus, shown in Figure 51, was developed to measure in an objective way the degree of proficiency with which a subject can coordinate the eyes, arms, and legs. In this test the subject manipulates airplane controls to direct a rectangular beam of light at a series of targets reached by following definite paths. The rudder bar is coupled to the lamp in such a way that it moves the beam of light horizontally from right to left. Forward and backward movements of the stick raise and lower the beam in the vertical plane, and right and left movements of the stick rotate the beam through an angle.

Both the total time required for a set of reactions and the errors (false moves) made as departures from the correct path were recorded. Each of the five subjects made 20,400 reactions in noise and the same number in quiet. The reaction in noise was slower by 5.4 per cent. Also, the number of errors committed in noise was greater by exactly the same proportion, 5.4 per cent.

Of all the motor-coordination tests employed,

this was the only one which showed a decrement in performance in the presence of noise. Other tests which appear to be closely related to this serial reaction-time test showed no effect, as will be seen below. A consideration of several possible explanations of this apparent paradox led to the belief that in the less intense noise the subjects made use of certain acoustic cues provided by the "click" of timing relays which controlled the apparatus. This possible artifact was not experimentally tested because of the press of time, and consequently the slight decrement obtained by this apparatus cannot be considered valid until further tests are made.

FIGURE 51. Coordinated serial reaction-time apparatus.

MUSCULAR TENSION

There is no doubt that noise causes in some subjects an increase in muscular tonus, but attempts to measure this effect by recording electric potentials in various muscles were often frustrated by artifacts of gross muscular movements. Voltages from the arms and legs were recorded under various conditions or activities. Subjects are apparently able to relax as completely in noise as in quiet, provided they try to relax, but when records are made at a time when the subjects are busy at another task, there is evidence of increased tension due to noise. But as already indicated, artifacts of movement too often obscure the records.

METABOLISM

Since the variation in metabolic rate from day to day and even during a single day is so large, it was not feasible to determine the effects upon metabolism of long exposures to intense noise. Shorter exposures repeated several times during the day indicate that for some subjects there is an increase in metabolic rate as a function of noise. In other subjects no consistent effects could be noted.

BREATHING

Breathing was recorded in the process of measuring metabolism and also under other experimental conditions. Noise definitely causes

some subjects to breathe more rapidly and less deeply, but here again variability and individual differences forbid generalizations.

Speed of Accommodation

The speed with which the focus of the eye can be changed from a near to a distant object, and vice versa, is apparently reduced by long exposure to noise. This effect was present in many of the experiments, but individual differences do not permit a simple statement of the magnitude of the reduction.

Saccadic Eye Movements

The speed with which the visual line of regard can be shifted through an angle of 37 degrees was measured for four subjects for more than 5,000 eye movements, but in only one subject was the speed of movement slower in noise than in quiet.

Body Sway

The ability of a subject to stand erect without swaying was measured by an ataxiameter. Although this performance was not tested intensively, preliminary results suggested that there exists no striking relation between steadiness and noise.

Hand Steadiness

The subjects showed wide individual differences in their ability to hold a small stylus in the center of a small hole (3-mm diameter) without touching the walls. The effect of noise was to produce a slight but statistically unreliable increase in steadiness. It is possible to attribute this improved performance to the ability of the airplane noise to mask other distracting stimuli and thereby "insulate" the subject from his environment. This "insulating" effect was also reported by the subjects who participated in the experiments on marksmanship. The results of the studies on marksmanship indicated that noise of the temporally continuous character employed tends to increase the accuracy of offhand shooting.

Reversible Perspective

When a person views an outline drawing of a figure such as a cube or a staircase he observes periodic shifts in perspective. The cube appears sometimes as seen from above and sometimes as seen from below. The subjects were asked to fixate a reversible figure and to press a key at the occurrence of each involuntary reversal. If there is any relation between the instability of this type of perception and noise it was in our experiment obscured by variability.

Dark Adaptation

The threshold of visual illumination was measured at the beginning and at the end of seven-hour periods of exposure to noise. These thresholds were actually about 0.05 log units higher than after a similar period in quiet, but this difference may not be attributable to the effect of noise on the subjects because a similar difference occurred at the very beginning of the seven-hour period. Additional research is needed to discover the reason for these results. Also there appeared a slight but insignificant effect on the visual threshold when the noise was turned on and off at approximately ten-minute intervals.

2.5.3 Tests on Which Noise Had No Effect

The following tests are those on which the subjects made the same score in noise as in quiet. The results of these tests are not to be considered negative in the sense of being inconclusive. They are positive in the sense that they prove that, under long exposures to intense sound, subjects are able to maintain their levels of performance on a wide variety of tasks.

Coordinated Serial Pursuit

The coordinated serial pursuit meter was developed as a task which partially simulates the requirements of instrument flying. The subject is required to adjust by means of airplane controls the position of the spot on a cathode-ray oscillograph and the position of the pointer on a direction meter in such a way as to compensate for deflections introduced by external circumstances. A view of this apparatus is shown in Figure 52.

A graphic record of each of the three dimen-

sions of movement (forward-backward and lateral on the oscilloscope and direction on a separate meter) is written on three Esterline-Angus recording galvanometers. In this way, a record is obtained which reveals not only the number or extent of the subject's failures, but also the nature of these errors.

FIGURE 52. Coordinated serial pursuit meter.

Subjects were able to maintain their performance on this test unimpaired in airplane noise. There was some indication in the verbal reports of the subjects that they maintain their performance by dint of greater effort. At least, they were aware of a tendency to compensate for the stress of the noise. If it was in fact a matter of compensation, it is possible that, if the pursuit meter were speeded up sufficiently, a significant difference between the performance in quiet and noise might appear.

SERIAL DISJUNCTIVE REACTION-TIME

Quite unimpaired in noise was the ability of the subject to press a key with the appropriate hand or foot, depending on which of four lights was illuminated in front of him. Each subject made 24,480 reactions in noise and the same number in quiet. Had there been a difference between noise and quiet of 0.6 per cent it would have been statistically significant. The actual difference was only 0.1 per cent.

FAST-SPEED PURSUIT ROTOR

Also unimpaired was the ability of a subject to follow with a stylus a small disk near the edge of a phonograph table revolving 99 rpm.

CARD SORTING

It was demonstrated early in the experiments that ability to sort 12 kinds of cards into 12 compartments is maintained as well in noise as in quiet. As soon as this was discovered the test was dropped from the procedure.

CODING TEST

The subjects were able to translate written

material into code, letter by letter, as rapidly in noise as in quiet.

Judgment of Distance

Noise had no effect on monocular judgments of distance as measured by the ability of the subjects to adjust the distance of a movable wire until it was as far away as a standard wire.

Conclusion

Consideration of these results leads to the conclusion that airplane noise, apart from the other stresses to which an aviator is subjected, has at worst only a slightly detrimental effect upon functions involving motor coordination, reaction-time, sensory perceptions, and certain mental functions. In most instances it can be positively demonstrated that noise has no effect, even after exposures lasting seven hours. Functions such as breathing, metabolism, and muscle tension are sometimes affected by noise, but what the effect will be seems to depend upon the kind of individual tested, and simple generalizations are not possible.

Nevertheless, all subjects seem to prefer not to work in an intense noise, and after a day under acoustic stress, they tend to report a subjective feeling of being more tired and irritable. They have a ringing in the ears and a temporary hearing loss to testify to the severity of the conditions under which they and the pilots of aircraft labor.

The most severe effects of noise are upon the ear itself. Speech communications are impaired and temporary losses in hearing are produced. The problem of stimulation deafness was later assumed under the Committee for Medical Research,[16, 17] while the studies of communication in noise were continued by NDRC Section 17.3.

2.5.4 Effect of Vibration on Visual Acuity

Ten subjects were tested for visual acuity under two conditions. Under one condition the subjects were seated on a stationary chair. Under the other condition the chair vibrated at a rate of 2,300 rpm. Variable-speed motors, driving eccentric loads, were used to generate vibrations similar to those encountered in airplanes. Visual acuity was measured by means of a grid of parallel lines whose separation and whose angle with the horizontal could be continuously varied.

In every subject vibration produced a considerable reduction in visual acuity. The average reduction for all subjects was 25 per cent. The reduction for horizontal lines was 5 per cent greater than for vertical lines. The reduction seems to depend upon the effective vibration at the subject's head. For a fixed vibration of the chair, this effective amplitude at the head appears to depend upon the body-build of the individual. Measurements showed an amplitude of vibration at the head equal to approximately 0.001 in.

Further study of the effects of vibration on psychomotor efficiency were planned, but a growing preoccupation with communication problems made it necessary to abandon the vibration research.

2.6 AURAL PROTECTIVE DEVICES: EAR WARDENS

Although most ears recover completely within a few hours from the impairment that follows the usual airplane mission or a day in a noisy factory, there is considerable indirect evidence that permanent hearing losses, particularly for high tones, may follow repeated exposures to acoustic stress. Protection of the ear against these effects can be accomplished by isolating the sources of the noise, by acoustical treatment of the surroundings, or by stopping the ears themselves against the sound. This last method is the most applicable in the usual military situation, and for this purpose the development of a satisfactory earplug was undertaken.

The earplug was designed to meet three requirements: acoustic insulation, comfort, and distortionless transmission of speech frequencies. Comfort can be ranked as a consideration equal in importance to acoustic insulation. The material must be soft and compliant enough to shape itself to the contours of the ear canal.

Also, an earplug intended for military use must not impede communication under the conditions in which the earplug must be worn. An earplug that reduces the transmission of noise also reduces transmission of speech sounds. Hence in situations where loud noises make the use of earplugs desirable, the earplug selected should attenuate all the frequencies of speech approximately the same extent. This insures against distortion of the speech by the protecting device while, at the same time, it reduces the whole complex of speech and noise to a level to which the wearer can more comfortably listen. At this more comfortable level, a given speech-to-noise ratio can result in improved communication.

These basic requirements were recognized when the initial NDRC project was undertaken in November 1940 in the Department of Physics, University of California at Los Angeles [UCLA].[11] The program of this project resulted in the development of the Type V-29 ear defender. The principles established by this work guided the subsquent development at the Psycho-Acoustic Laboratory, where the NDRC Project was continued. Of the resulting models, the ear warden Type V-51R semed to offer the best combination of desirable properties and this model was consequently recommended for Service use.

2.6.1 Design

The body of the V-51R ear warden is elliptical in cross section rather than round, since this shape has been found to approximate more closely the shape of the average ear canal. Adapting the earplug to the shape of the ear canal aids in securing for the wearer greater comfort and better protection. A longitudinal section of the earplug is shown in Figure 53.

The bell-shaped flange on the forward end of the ear warden aids in establishing a good seal. The septum is a dome-shaped structure, convex outward, situated at the beginning of the stem which holds the bell-shaped flange. This stem is hollow for greater flexibility. The safety tab perpendicular to the axis of the main body aids in sealing the earplug in the ear canal and prevents the earplug from being inserted too far. The removal pad makes it easy to withdraw the earplug.

The theory of earplugs has not been fully formulated. This is explained in part by the extreme complexity of the acoustic system

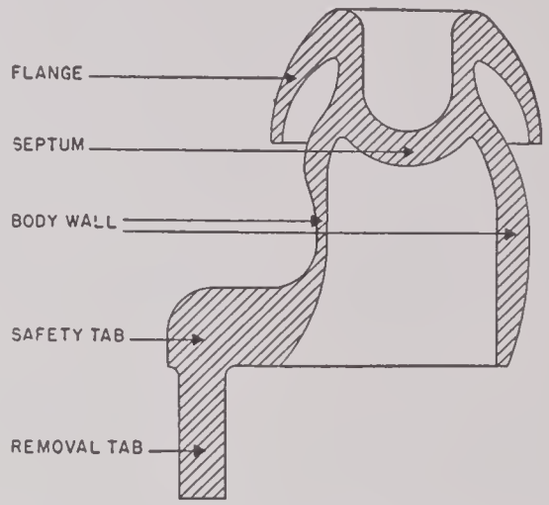


FIGURE 53. Cross section diagram of the V-51R ear warden.

which results from the introduction of an earplug into the ear canal, and in part by the fact that the experimental determination of the various acoustic quantities requires elaborate experimental apparatus. In the V-51R ear warden the bell-shaped flange and the thin, compliant body wall are designed to reduce leakage to a practical minimum. Shaping the septum and related parts in such a manner that a high stiffness factor obtains minimizes transmission through the earplug. The small mass of the ear warden, together with the fact that the body wall offers a large frictional area of contact, tends to obviate any tendency of the ear warden to oscillate as a whole.

From a theoretical point of view, any of the variables—mass, stiffness, or frictional resistance—may be used as a means of preventing the transmission of sound through a device, and the particular combination of these properties determines the frequency characteristic of the transmission loss. In particular, it is important to emphasize that both stiffness and friction can serve as the principal impedance and to minimize the importance sometimes attached to mass.

The earlier work at UCLA produced several designs involving mass as the principal im-

pedance. In order to secure comfort for the wearer, the mass was incorporated in a compliant structure, and, as might be expected, some of the devices behaved as tuned circuits, with a consequent loss of insulation. The V-51R ear warden, like the V-29 which preceded it, was designed to present mainly a stiffness impedance to the incoming sound. In such a device, adequate impedance to frequencies between 100 and 1,000 c can be had from stiffness and friction, provided the resonant frequency of the earplug in its suspension in the ear canal is high, and provided the impedance at the resonant frequency is largely dissipative.

Use of wardens in aircraft suggests a special problem of design. There is evidence that equalization of the pressure in the ear canal and in the middle ear is not readily accomplished by some people if the changes in altitude are very large or very rapid.[22] In a minority of ears inflammation of the ear canal developed. A special slow leak in the septum of the earplug is a possible answer to this difficulty.

2.6.2 Material

The ideal material for the manufacture of earplugs would be at once malleable and resilient. Its physical characteristics would not be affected by aging, by exposure to temperature extremes, or by the action of earwax, antiseptics, and cleansing agents. Since the snugly fitting earplug is maintained in intimate contact with the lining of the ear canal, the plug must be made from a nontoxic material. The material must be compliant not only for comfort but also in order that a good seal may be established. The material should not be permanently deformed as the result of wear. From the practical point of view of economical production and distribution, the material must be easily molded. The percentage of earplugs defective because of the failure of the material to mold, or from stripping the molds of the earplugs, should be small.

The most satisfactory results thus far obtained at the Psycho-Acoustic Laboratory were achieved with polymerized chloroprenes, e.g., neoprene, which, when formulated with a maximum of nontoxic plasticizer, yield ear wardens that are soft, reasonably resilient, and markedly resistant to the action of earwax. Promising also are the polyvinyl-chloride-copolymers, e.g., Vinylite. The chief technical problem with this material centers about the choice of plasticizer, since some plasticizers are irritant, some are ineffective at low temperatures, and some are generally extracted from the warden by contact with earwax.

2.6.3 Acoustic Insulation

The amount of acoustic insulation afforded by an earplug can be evaluated either by the threshold technique or by the method of loudness balances. By the threshold technique the observers make measurements monaurally, using the right ear. The left ear is carefully blocked off, and the earplug to be tested is inserted by the observer into his right ear. He is then seated in a chair in an anechoic room with his right ear toward a loudspeaker one meter away. The observer adjusts the intensity of the test tones until he determines his threshold of hearing.

In order to evaluate the amount of insulation for any frequency, it is necessary to determine the amount by which the open-ear threshold is shifted when the earplug is placed in the ear. Two series of measurements are therefore necessary. One threshold measurement is made at each frequency for the open ear and another at each frequency after the ear has been plugged. The difference between the two intensities is the amount of insulation afforded by the earplug.

In Figure 54 is presented a comparison of tan neoprene, black neoprene, and Vinylite models of the V-51R ear warden. For all practical purposes the three curves are identical. Acoustic insulation values range from approximately 26 db at the low frequencies to 40 db at the high frequencies.

In addition to the V-51R ear warden, the following earplugs were also examined at the Psycho-Acoustic Laboratory (see Figure 55).

A. Baum's Mega eardrum protectors. Nipple-shaped, soft rubber capsules, bearing four

longitudinal grooves. Each pair of protectors is connected by a long, stout thread. One size.

B. Nods noise mufflers. A cylinder of latex sponge impregnated through about two-thirds

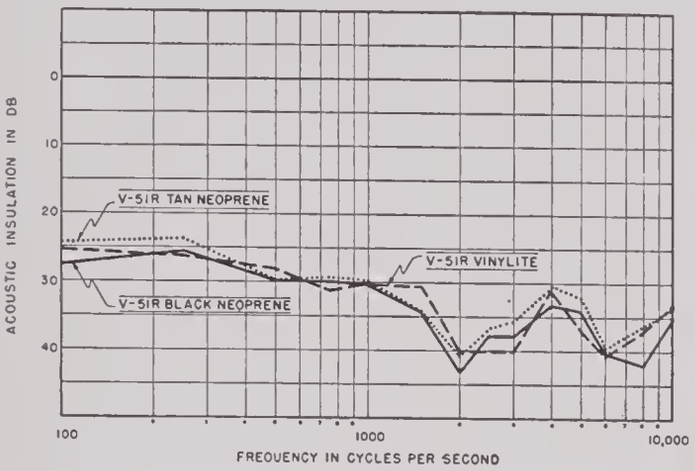


FIGURE 54. Comparison of acoustic insulation for three models of the V-51R ear warden. Data obtained by threshold technique.

of its length with antiseptic wax. Kneaded with the fingers to proper size.

C. Olygo noise absorbers. A cylinder of cotton impregnated with antiseptic wax, kneaded with the fingers to proper size.

D. Flents antinoise ear stopples. A cylinder of cotton impregnated with antiseptic wax, kneaded with the fingers to proper size.

E. Nelson's ear stoppers. A hard spherical globule of colorless plastic terminated in a T-shaped handle which facilitates insertion and removal. Three sizes.

F. Sepco safety ear protectors. A bullet-shaped capsule made of rubber, or rubber-like synthetic material, is filled with flocculent rubber enclosed with a plug of rubber sponge. An external flange on the cylindrical body of the plug limits the depth of insertion in the ear canal. Near the rounded tip of the plug there is a minute aperture or puncture. Five or more sizes. Used with antiseptic lubricant jelly.

G. SMR ear stoppers. These are similar to the Sepco safety ear protectors. Five or more sizes. Used with antiseptic lubricant jelly.

H. Mallock-Armstrong ear defenders. These are hard, black, plastic plugs. They have a channel through them which is terminated at the outer end by a cellophane-like diaphragm placed between two layers of fine wire screen. The flare at the base limits the depth of insertion. Five sizes. Very similar devices have been found on captured German soldiers and in Japanese military stores. The intent of the design is an acoustic valve.

I. MSA ear defenders. A capsule of gum rubber encloses a metal insert of the shape illustrated. Three sizes.

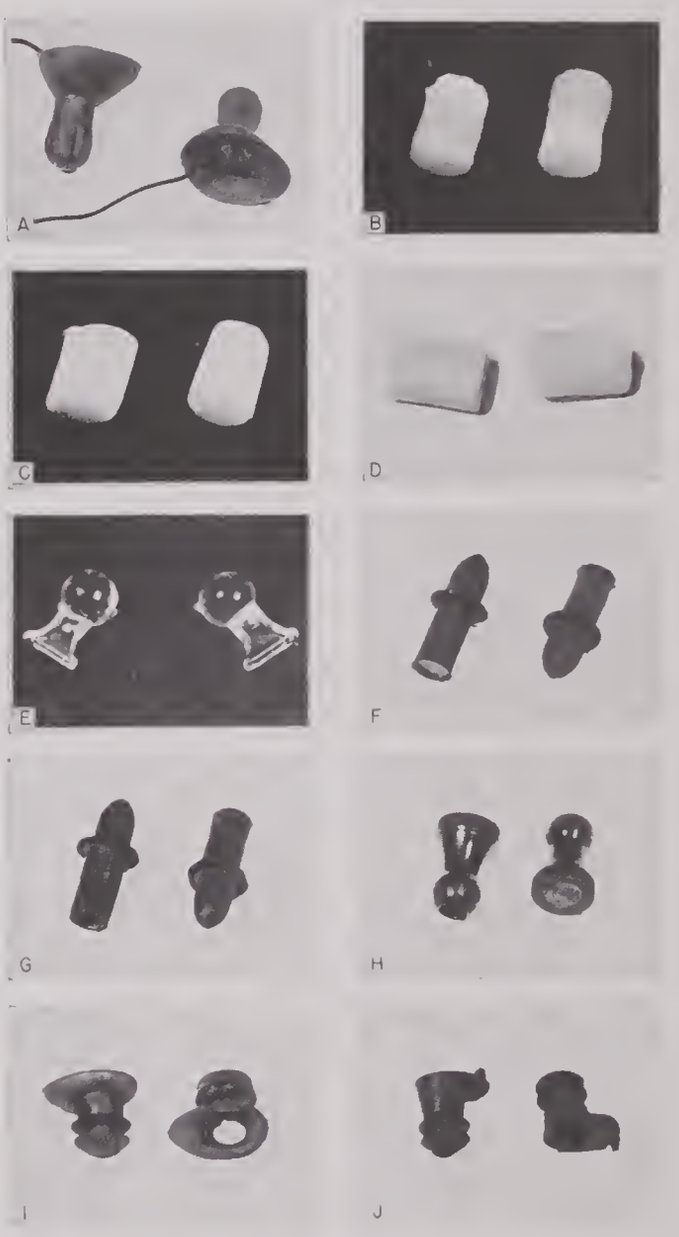


FIGURE 55. Some earplugs evaluated for acoustic insulation.

J. NDRC ear wardens (Type V-29R). These are capsules of soft neoprene. Like the MSA ear defenders, they have two concentric flanges. A safety valve is provided to limit the depth of insertion. A tab at the base facilitates removal. Three sizes.

The resulting insulation data for these ten earplugs are given in Figure 56. For purposes of comparison, the data include the results obtained by the common expedient of stuffing the ear canal with cotton. All the earplugs tested,

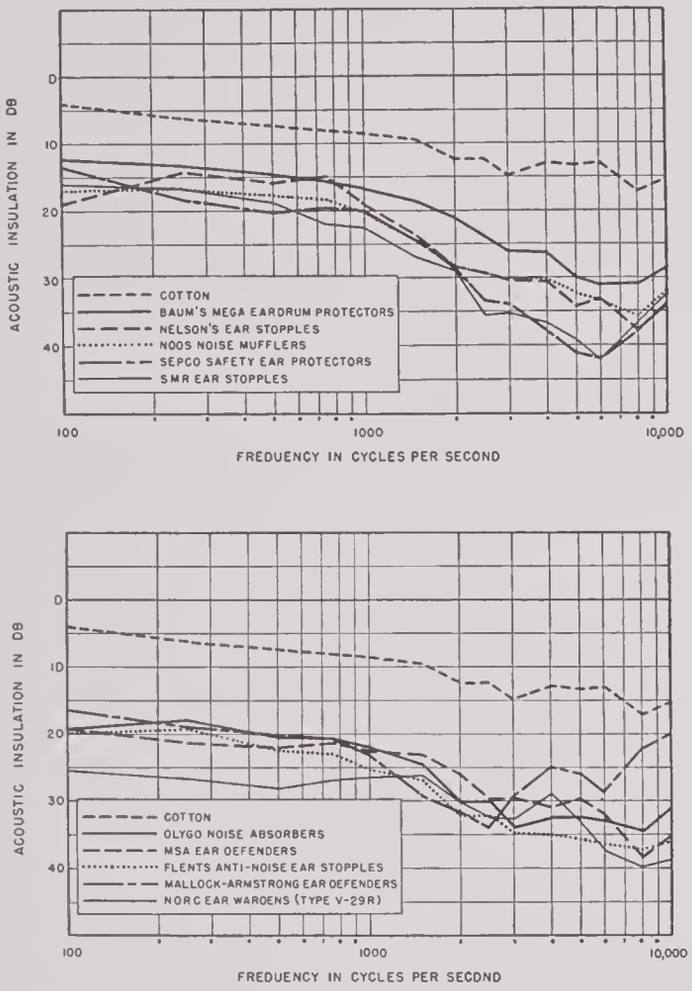


FIGURE 56. Acoustic insulation provided by the earplugs pictured in Figure 55. Cotton is included for purposes of comparison.

with the possible exception of Baum's Mega eardrum protectors, have good acoustic insulation for the range of frequencies from 2,000 to 10,000 c. Over the low-frequency range, where protection is hard to attain, the various earplugs differ considerably in the amount of acoustic insulation afforded.

The validity of the threshold method of determining acoustic insulation has been challenged from time to time. One way of testing the insulation at higher levels of noise is by the method in which a loudness balance is obtained between sounds introduced into the two ears by two separate earphones. The observer controls the level of the test signal in one earphone until the sounds in the two ears appear equally loud. After the balance is obtained, an earplug is inserted into one of the ears, and the loudness is again balanced. The amount, in decibels, by which the intensity of the sound must be increased in order to appear equally loud is taken as a measure of the insulation afforded.

The results of the loudness-balance measurements of the insulation of the V-51R ear wardens are presented in Figure 57. In addition to the curve for loudness balance, a curve of insulation measurements for the same observers

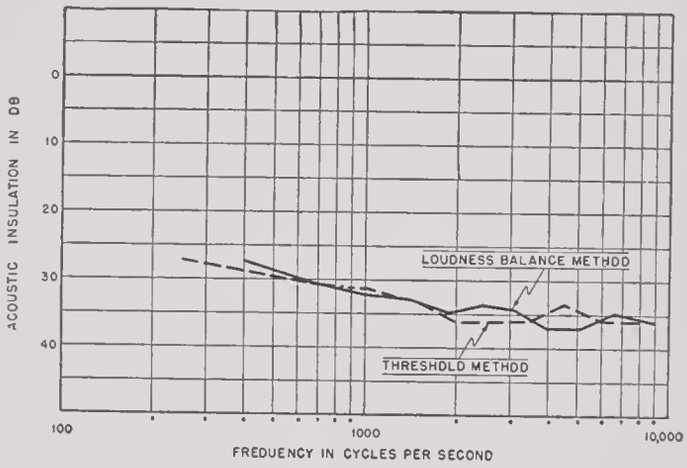


FIGURE 57. Comparison of acoustic-insulation curves obtained by the loudness-balance and threshold methods.

using the threshold technique is shown for comparison. The latter curve shows that the magnitude of acoustic insulation is essentially the same at high intensities as at low intensities of noise.

2.6.4 Articulation Measurements

The results of articulation tests show that with most Service interphones, operating at normal amplifier gains and with normal voice levels, the wearing of ear wardens does not impair the reception of speech. On the contrary, reception of speech may be improved. Figure 58 shows the results obtained for articulation tests performed in the quiet and with various amounts of ambient noise introduced into the test room. Speech of weak intensity, in the quiet and in noise levels below 75 db, is not heard as

readily by listeners wearing ear wardens. In the quiet, however, and for all noise levels, speech slightly above conversational level (75 to 80 db) is heard equally well by listeners with and without ear wardens. In noise levels above 75 db the intelligibility of speech, as is shown by Figure 58, is better with ear wardens. A signal that is audible to the open ear in a noise higher than 75 db is equally audible or more audible when ear wardens are worn. In the noise levels usually encountered in submarine engine rooms, tanks, and airplanes, ear wardens should improve speech communication.

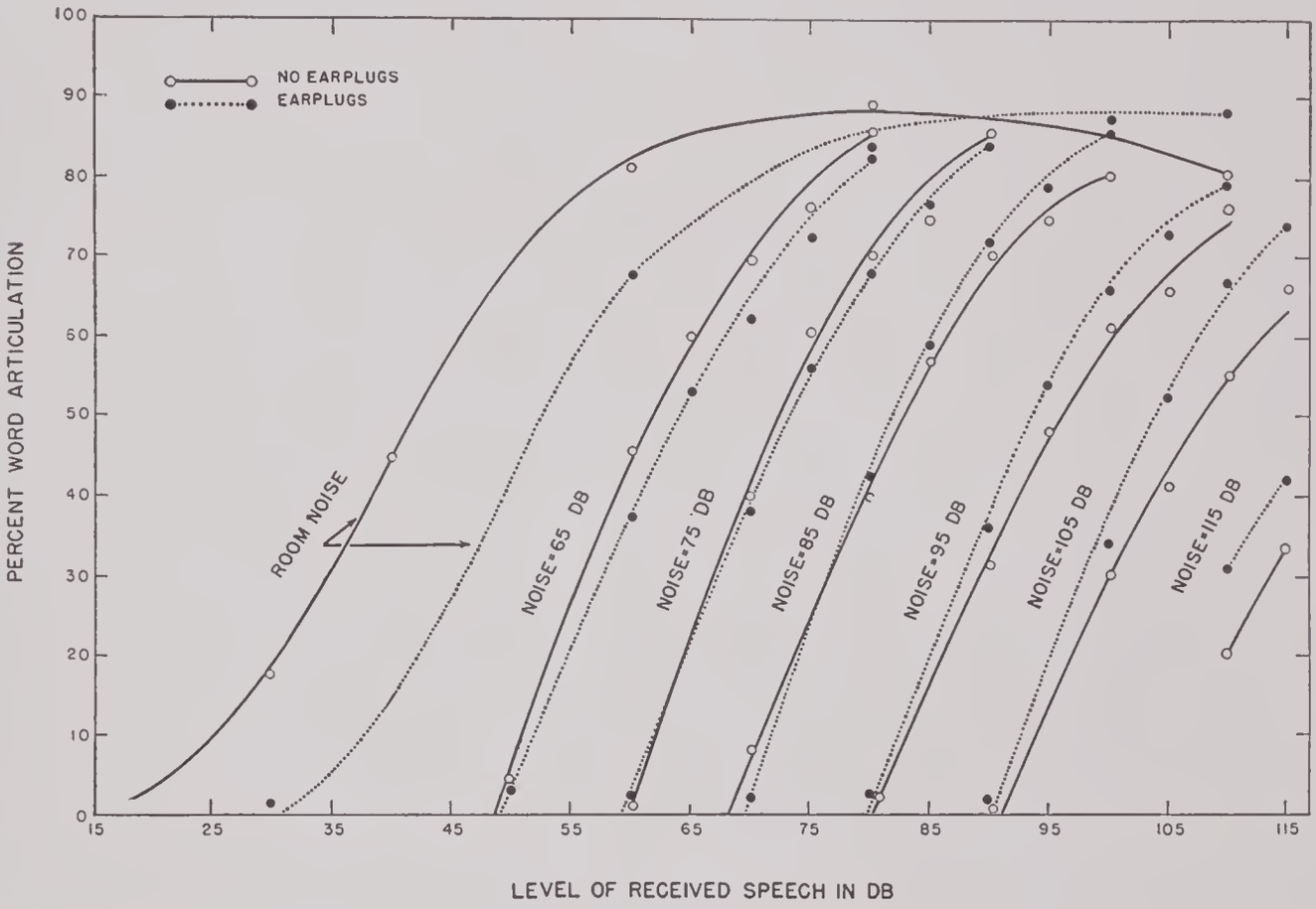


FIGURE 58. The relation between articulation and speech level with noise level as the parameter. In loud noise ear wardens improve articulation.

2.6.5

Blast Exclusion

The disastrous effects on unprotected ears of the discharge of nearby artillery or the close burst of a large caliber shell are well known. Severe exposure in front of a large gun or really close escape from a shell burst or land mine ruptures the eardrums and may also disrupt and cause permanent degeneration of the delicate sensory structures in the inner ear.

The problem of protection against gun blasts differs slightly from that of protection against continuous unwanted noise. A blast may be described in physical terms as a single sound-wave, a "transient," of unsymmetrical form. The major wave is a single violent but brief wave of positive pressure amounting to many pounds to the square inch and greatly exceeding any pressures reached by sustained sounds. The positive pressure wave is followed by a longer but less intense wave of negative pressure, and by smaller secondary waves. Fortunately, the ear can fairly well tolerate intense single waves which would rapidly cause severe damage if they were repeated several hundred times a second, as in a steady tone. But there is a limit beyond which a blast becomes strong enough to produce temporary deafness, particularly if it is a sharp blast with a steep initial wave front.

In order to protect the ear against blast waves the pressure wave must be attenuated, and the problem is similar to the problem of attenuation of any other noise in the audible range. The same considerations of comfort, wearability, simplicity, and cleanliness that apply to defenders against continuous noise apply equally to protectors against blasts. It is desirable, however, that the earplugs should not be moved bodily inward by the pressure wave.

The NDRC ear wardens, Type V-51R, were originally designed for protection against continuous noise rather than against blasts. They nevertheless fulfill most, if not all, of the obvious requirements of blast protectors. The design of the ear warden, with external safety tab and a large flange in close contact with the wall of the ear canal, effectively prevents its being driven into the ear canal by the pressure wave of a blast.

Testing the ear warden in the presence of gunfire in the field is not only the most practical and economical but also the most conclusive kind of test of its ability to protect the ear from blasts. Accordingly, a number of field tests were arranged. Through the courtesy of the Field Artillery Board, members of the Psycho-Acoustic Laboratory were privileged to attend an informal test of the V-51R ear warden in the immediate vicinity of some of the heavier armament at Fort Bragg, North Carolina. After the firing, a short questionnaire was given to all men participating in the field test. On the basis of these results, the Field Artillery Board concluded that the V-51R ear wardens "are suitable as replacements for cotton or cotton waste by members of the firing battery." After similar tests, the Chemical Warfare Board also concluded that ear wardens are superior to cotton in protecting the wearer against blast effects, and that communication is not impaired when ear wardens are worn.

Following formal or informal trials of the wardens, and examination of a considerable number of commercial earplugs, procurement of neoprene or Vinylite wardens was undertaken in behalf of the United States Navy, the Royal Navy, the Royal Canadian Navy, the United States Army Air Forces, and the Chemical Warfare Service, U. S. Army.

Chapter 3

SOME CHARACTERISTICS OF THE HUMAN EAR

ONCE IT had been determined (see Section 2.5) that the only major effect of loud noises on man was interference with normal auditory perception, the research endeavors of the Electro-Acoustic and Psycho-Acoustic Laboratories were directed toward the more pressing problems of communication. Since speech-communication systems usually terminate in human ears, it will facilitate discussion in the following chapters if some of the fundamental facts about the ear are here made explicit. (Several comprehensive treatments of hearing are available.)[3, 5, 8]

A schematic drawing of the ear is shown in Figure 1. The visible portion of the ear (auricle or pinna) is a fibrocartilaginous structure leading to the external auditory canal. This canal is from 2.0 to 2.5 cm long, approximately 0.7 cm in diameter, with a volume of about 1.0 cu cm and an area at the opening of approximately 0.4 sq cm. Pinna and external canal together are referred to as the outer ear.

The canal is closed at its inner end by the cone-shaped eardrum (tympanic membrane). Internal to this membrane is the middle ear, enclosing a cavity of 1 to 2 cu cm volume and containing the three small bones (ossicles) which transfer the agitations of the eardrum to the inner ear. Pressure on the two sides of the membrane is equalized by the Eustachian tube, which opens during the acts of swallowing or yawning.

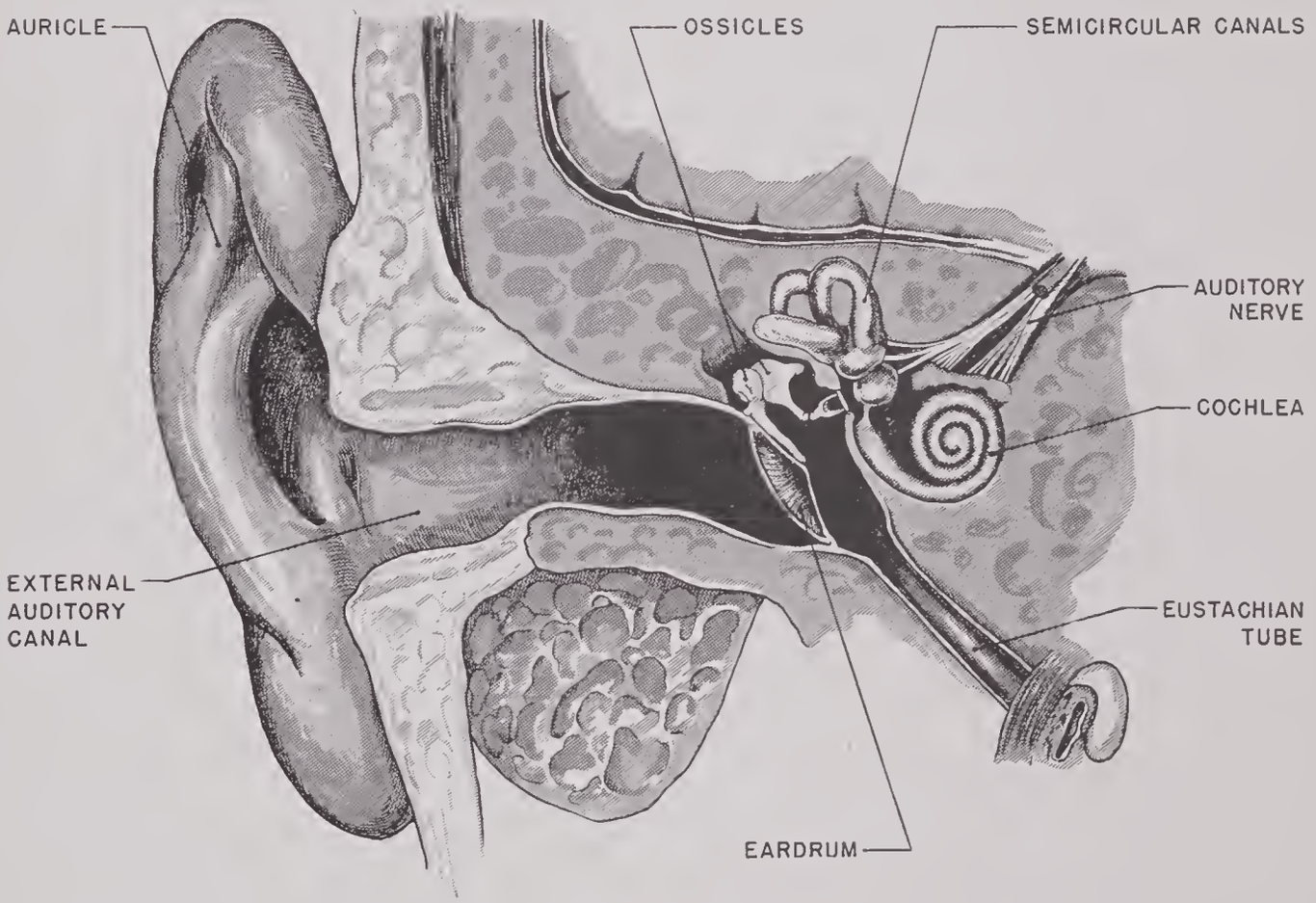


FIGURE 1. Schematic drawing of the ear.

The inner ear (cochlea and semicircular canals) is filled with liquid and surrounded by bone. Acoustic stimulation is transmitted via the ossicular chain to the liquid of the cochlea which, in turn, causes movements of a membrane (basilar membrane) supported in the

liquid. The auditory nerve endings are distributed along this basilar membrane. Presumably, different nerve endings respond to different frequencies, depending upon their location along the membrane, and in this way the inner ear can operate as an acoustic analyzer.

Thus the acoustic stimulus is transmitted through the auditory canal to the drum and is converted into mechanical movement of the bones of the middle ear. This movement leads to hydrodynamic action in the inner ear which results, finally, in the electrochemical event known as a nerve impulse. Nerve impulses travel along the eighth cranial nerve to the central nervous system and there, by ways not understood at present, lead to sensations of hearing. The actual course of events can be followed with some confidence as far as the tympanic membrane; beyond that, the details become controversial.

3.1 SOUND-PRESSURE DISTRIBUTION IN THE AUDITORY CANAL

The physical stimulus acting on the ear can best be specified by measuring the sound pressure at the eardrum. Resonance effects in the ear canal and diffraction effects around the head are thereby taken into account. An experimental study[17] was made with the object of determining these effects and also the variations, as a function of frequency, of the sound pressure along the auditory canals of a number of subjects (male and female) placed in a plane progressive sound field.

A small, flexible, probe microphone was inserted at various positions along the length of the auditory canal, and the sound pressure determined as a function of position and frequency. The subjects were placed in the sound field of a loudspeaker in a room free from acoustic wall reflections (anechoic chamber). The free-field sound pressure at the subject's location was also determined. The ratio of the sound pressure at the eardrum to the free-field sound pressure was taken to be a measure of the combined obstacle effect of the head and resonance effects in the ear canal. In addition, sound-pressure measurements were made at the entrance of the ear canal and at a point about halfway down its length. These measurements serve, to some extent, to separate the resonance and diffraction phenomena and to explore the pressure distribution in the auditory canal itself. The results are plotted in the form of pressure ratios as a function of frequency for values of azimuth, ϕ (the angle between the loudspeaker axis and the plane of symmetry through the subject's head), ranging from 0 to 90 degrees.

The subject was seated in a chair 6 ft away from the loudspeaker, and his head was secured by a head clamp mounted on the back of the chair. The clamp was designed (see Figure 2) to permit the fixing of the subject's head with a minimum of discomfort. This is necessary in order to make possible the insertion of the probe microphone into the auditory canal with a minimum of danger due to accidental shifts in head position. The microphone used in these tests consisted of a Western Electric Type 640-AA condenser microphone to which a probe tube of small diameter was coupled by means of suitable coupling rings (see Figure 2). About

FIGURE 2. Sound-pressure measurement in the auditory canal.

half the total length of the tube consisted of brass tubing; the rest was made up of flexible plastic tubing. It is this feature, incorporating sufficient softness and flexibility, which is essential for the application at hand.

To permit easy and accurate insertion of the microphone into the ear canal, the microphone holder was mounted in a special adjustable carriage (see Figure 2). The carriage is designed to permit easy and free adjustments in three orthogonal directions. The chief problem involved in the measurements was the insertion of the probe tube into the ear canal to the point where its open end would be "at the eardrum." This was achieved by thrusting the probe gently into the canal by means of the lateral motion adjustment of the holder. The subject was instructed to advise the experimenter as soon as (1) any pain or (2) any auditory sensation was experienced. Experience has shown that when the eardrum is contacted, an auditory sensation is experienced which has the character of a dull thump. After the subject reported contact with the eardrum, the probe was withdrawn about 1 mm and measurements were begun.

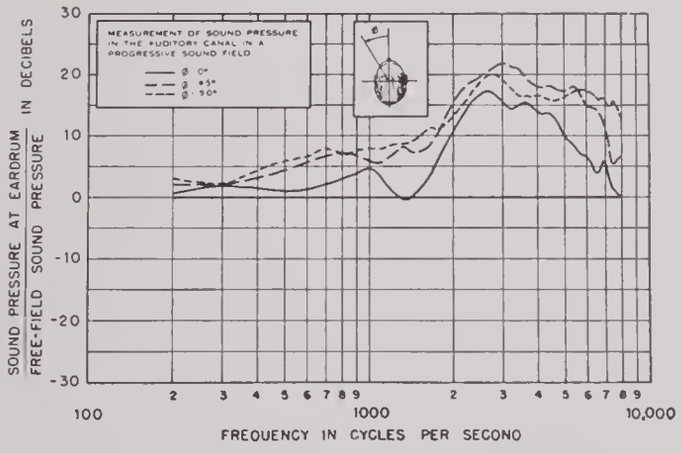


FIGURE 3. Ratio of sound pressure at the eardrum to sound pressure in the free field at the center of observers' head. Average of 6 to 12 male left ears.

Figure 3 shows, for six to twelve male subjects, the average ratio of the sound pressure at the eardrum to the free-field sound pressure, plotted as a function of frequency. It is evident from the figure that the human ear is an effective acoustic "amplifier," with maximum amplification of the order of 20 db near 3,000 c. The effect is somewhat dependent on azimuth, and is markedly a function of frequency. Figure 4 gives an estimate of the variation with azimuth in the form of a polar diagram. For the range of azimuths investigated, the maximum pressure

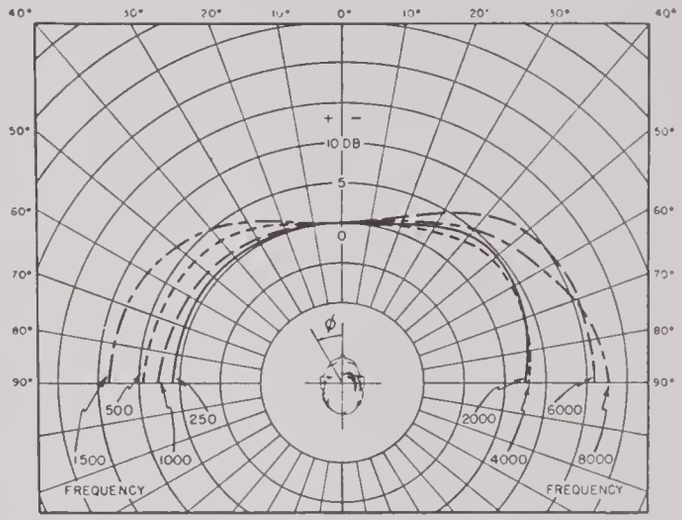


FIGURE 4. Variation of sound pressure at the eardrum with azimuth ϕ, for constant free-field sound pressure. Average of 6 to 12 male ears.

ratio is obtained near $\phi = 90$ degrees, where the subject has his ear turned toward the sound source.

Figure 5 shows the pressure distribution in the ear canal itself, expressed in terms of the

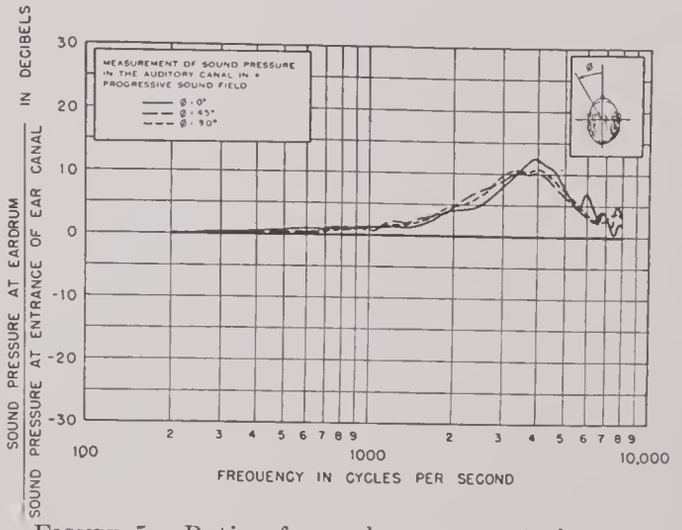


FIGURE 5. Ratio of sound pressure at the eardrum to sound pressure at the entrance of the auditory canal. Average of 6 to 12 male left ears.

ratio of the sound pressure at the eardrum to the pressure at the entrance of the canal. A resonant peak of about 10 db is reached at about 4,000 c, quite independent of azimuth. It can be shown[17] that diffraction around the head and pinna accounts for only the smaller part of the

total acoustic amplification. Resonance in the auditory canal contributes most of the pressure increase at the drum. A few measurements on female subjects indicated no essential difference in the acoustic behavior of the ear canal as compared with the measurements made on male subjects.

3.2 THE SENSITIVITY OF THE EAR

The absolute sensitivity of the ear is defined as the minimum sound pressure necessary to evoke an auditory experience. The point at which the sound pressure is measured must be specified in every case. As was pointed out in the preceding section, the sound pressure at the eardrum usually exceeds the free-field sound pressure producing it. Hence the thresholds will be different if expressed in terms of minimum audible pressure at the eardrum [MAP] or minimum audible free-field pressure [MAF].[6] In particular, the MAF is lower than the MAP produced by an earphone worn by the observer.

The differences between MAP and MAF found[6] at medium and high frequencies can probably be explained on the basis of the probe-tube measurements described in the preceding section. No entirely satisfactory explanation, however, has yet been offered of the difference of about 10 db found[6] at frequencies as low as 100 c.

The sound pressure at the threshold of audibility varies considerably for different frequencies (see Figure 6). At its best the ear is sensitive to frequencies between 20 and 20,000 c, but the region of maxium sensitivity is for frequencies between 1,000 and 4,000 c. At these frequencies the MAP approaches the reference level of 0.0002 dyne cm^2 adopted by the American Standards Association.[11]

3.2.1 Difference between Monaural and Binaural Sensitivity

It has been argued[7] that the binaural threshold is equal to the threshold of the more sensitive of the two ears. When the two ears differ substantially in sensitivity this argument applies, but experiments at the Psycho-Acoustic Laboratory indicate that, when both ears have approximately the same sensitivity, the binaural threshold for pure tones is about 3 db lower than the monaural.

A lower binaural threshold was obtained under two experimental conditions. In the first experiment, the thresholds for single ears were compared with the binaural threshold obtained with the same input to both earphones. In order to eliminate any effects of small differences in the sensitivity of the two ears, a second experiment established the two monaural thresholds and then determined the binaural threshold with the input to the two earphones adjusted separately to each of the monaural threshold values. In this way the sensation level (see Section 3.3.2) for both ears was made the same. This latter method is superior for estimating the difference between monaural and binaural sensitivity, although both methods indicated that the binaural threshold for pure tones in quiet is lower than the monaural threshold.

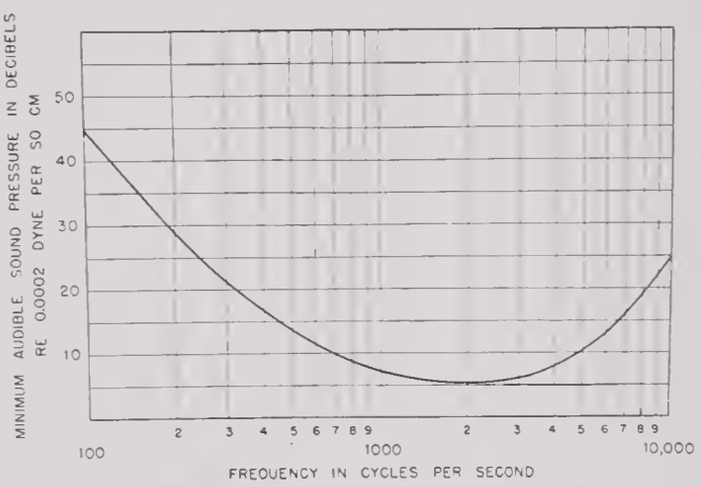


FIGURE 6. Monaural threshold for the average normal ear.

3.2.2 Altitude Effects on Sensitivity

The monaural threshold of hearing was determined at sea level and at 35,000 ft for fifteen subjects listening with earphones.[13] When corrections were made for the effects of the reduced ambient pressure on the earphone response and on the cushion leaks, no change in the auditory threshold was found.

3.3 PITCH AND LOUDNESS

It is commonly assumed that a direct, linear relation exists between apparent pitch and frequency and between apparent loudness and physical intensity. Experimental evidence shows that such assumptions are unjustified.

3.3.1 Relation of Pitch to Frequency

When listeners are instructed to adjust the frequencies of pure tones until they seem to be separated by equal pitch distances, the resulting scale of apparent pitch does not coincide with the scale of musical intervals. The *mel* has been chosen as the unit of apparent pitch and is so defined that a frequency of 1,000 c produces a tone of 1,000 mels.[10] A tone of 2,000 mels, for example, is twice as high in pitch as one of 1,000 mels and is produced by a frequency of 3,120 c. The pitch scale is shown in Figure 7, where pitch in mels is plotted as a function of frequency.

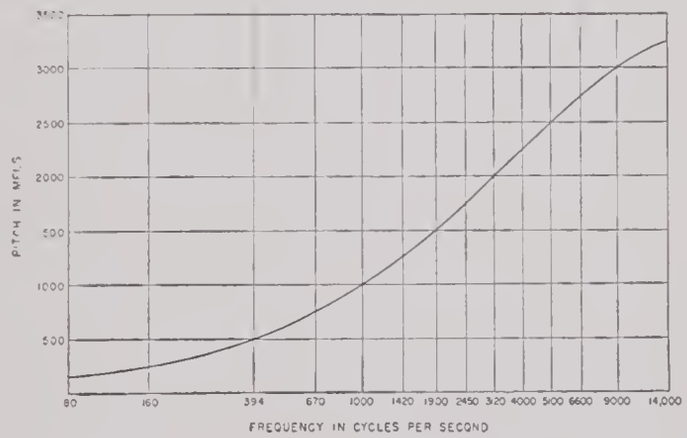


FIGURE 7. Pitch in mels as a function of frequency. Equal mel intervals represent subjectively equal pitch distances.

There is reasonably good agreement between this scale of pitch and the function which relates the position of maximum agitation on the basilar membrane to the frequency of the acoustic stimulus. The hypothesis which follows naturally from this agreement is that equal pitch distances represent equal distances along the basilar membrane. The scale can also be used to predict the sensitivity of the ear to small changes of frequency (differential sensitivity to frequency). A third correspondence appears between the pitch function and certain calculations used at the Bell Telephone Laboratories[12] for predicting the results of articulation tests. According to these calculations, equal pitch distances make approximately equal contributions to intelligibility.

The various suggested hypotheses have not been fully and rigorously tested, however, and the final unification of these several aspects of the auditory function awaits further research.

3.3.2 Relation of Loudness to Intensity

The physical intensity of a sound is usually expressed in decibels referred to some arbitrary value. The reference value determines the nature of the scale, and three different kinds of reference levels are commonly used. The *intensity level* of a sound is the number of decibels that the intensity of a free, progressive sound wave is above the arbitrary reference intensity. On this scale, 0 db $= 10^{-16}$ watt/cm^2 $= 0.0002$ dyne/cm^2. The *sensation level* indicates the number of decibels which the sound must be attenuated to reach the listener's threshold of hearing. The reference value in this case is the threshold, and conversions can be made from sensation level to intensity level when the threshold value is known.

The *loudness level* is the intensity level of a 1,000-c tone which is judged to be of the same apparent loudness as the given sound. The *phon* is the unit of loudness level. A loudness contour, for example, is given by all the tones which are equal in loudness and have the same value in phons. The magnitude of the psychological difference between one loudness level and another is not specified, however. Experiments have determined this scale of loudness, and the *sone* has been suggested as the psychological loudness unit.[8a] The sone is so defined that a tone of 1,000 c at a loudness level of 40 db produces a psychological experience having the magnitude of one sone. On this scale, for example, a tone at a loudness level of 70 phons is reduced to one-half its apparent loudness if the intensity is reduced by approximately 10 phons.

3.4 MASKING

For purposes of military communication in the presence of noise, one of the most important characteristics of the ear is its ability, as an analyzer, to respond selectively to certain components of the total acoustic pattern and to ignore others. This selective mechanism can be studied in its simplest form when two pure tones are introduced into the ear and the listener is asked to report the presence or absence of one or the other of the tones[1] (see Section 8.1.1). In such experiments it is found that the ear is not a perfect analyzer, but that some tones obscure the perception of others. The extent to which the signals interfere is called the *masking* of the one tone by the other.

The term masking is customarily defined as the shift in the threshold of hearing for pure tones which is produced by the introduction of an extraneous interference. The amount of masking is then measured in terms of the increase in the intensity of a tone that is required to make it heard in the presence of the interference. This notion can be readily extended to include speech as well as tones. The introduction of an interfering noise makes it necessary to raise the speech intensity in order for it to be understood. The necessary increment in the speech intensity is taken as the measure of the masking produced by the noise. If the *masked threshold* is defined as the sound pressure necessary to make a pure tone just audible in the presence of an interfering sound, then the amount of masking at that frequency is taken as the difference in decibels between the quiet and the masked thresholds. It is sometimes convenient to plot the absolute threshold in quiet as a straight line and to indicate the masking at each frequency in decibels relative to this line. Such a plot of masking against frequency is referred to as the *masking audiogram* for the interference used.

3.4.1 Masking of Tones by Noise[16]

In order to determine the masking effects of random noise on pure tones, four experienced listeners with normal hearing were tested individually in a quiet, electrically shielded room. Tones and noise were presented monaurally, and a frequency calibration for each listener was made with a small probe tube placed about 2 mm from the face of the earphone. The subjects determined their thresholds by adjusting an attenuator which controlled the intensity of the tone. Thresholds were obtained for 16 frequencies between 100 and 9,000 c for quiet conditions and for eight levels of masking noise. The eight noise levels varied from a sensation level of 20 to 90 db. Frequency analysis of the noise spectrum showed that the corresponding spectrum levels (level per cycle) of the noise varied from approximately —10 to 60 db.

When the masked thresholds are corrected for the frequency response of the earphone, the idealized masking contours shown in Figure 8

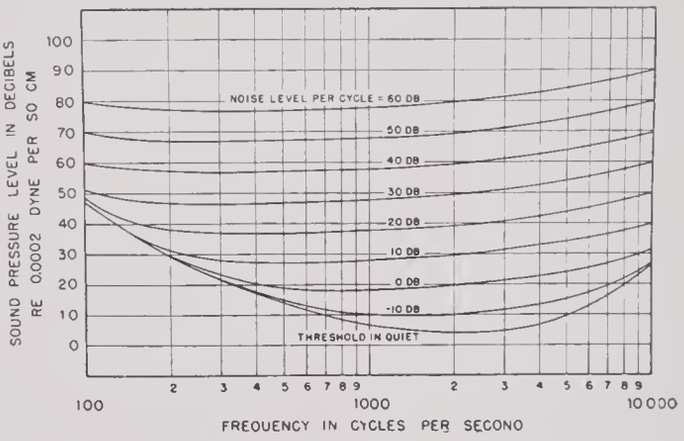


FIGURE 8. Masked-threshold contours. These curves show the monaural thresholds for pure tones when masked by various levels of white noise having uniform energy per cycle.

are obtained. The contours tend to be parallel, and are separated by intervals of approximately 10 db, corresponding to the intervals between the masking noise levels. The functions for the four highest noise levels are very similar. The ratio between the masked threshold of a given tone and the level per cycle of the noise at that frequency is the same for these top four contours, and in Figure 9 the average difference between the signal level and the level per cycle of the masking noise is plotted against frequency. The points represent the values obtained from the present experiment; the smooth curve is from data obtained at the Bell Tele-

phone Laboratories.[12] Thus we see that in order to be just audible, a tone of 1,000 c must be 18 db above the level per cycle of the noise.

The function in Figure 9 is sometimes taken as defining the "critical bandwidth" of a masking noise. Presumably it reveals what band of frequencies in a flat masking noise is effective

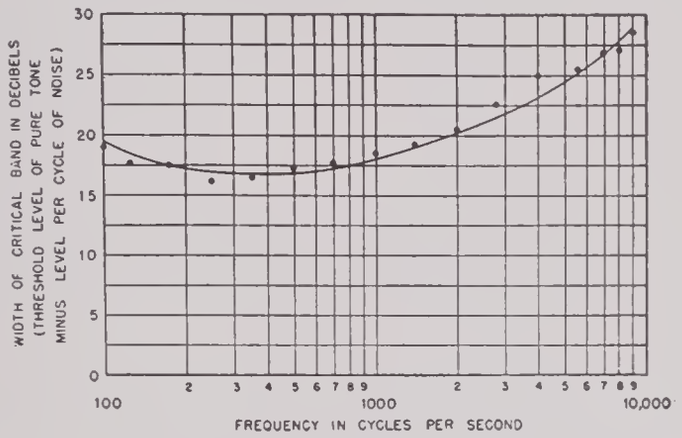


FIGURE 9. Ratio between the monaural masked threshold of a pure tone and the level per cycle of the masking noise measured at the frequency of the pure tone.

in masking a pure tone located at the center of the band. This interpretation rests on the assumption that the total energy required to mask the pure tone is equal to the energy of the tone itself. Since the energy in a band of white noise is proportional to the width of the band, it can be determined from Figure 9 how wide the band of noise would need to be in order for its total energy to be equal to that of a given pure tone at its masked threshold. For example, at 1,000 c the level per cycle of the masking noise is 18 db below the masked-threshold level of the pure tone, and since 18 db corresponds to an energy ratio of 63 to 1, the band of frequencies having a total energy equal to that of the 1,000-c tone would be 63 c wide.

The curve of Figure 9 is of further theoretical interest[9] because of the rather close agreement between the widths of the critical bands (expressed in cycles per second) and the widths of equal *pitch* intervals 50 mels wide (see Section 3.3.1).

These data can be considered from another point of view. It is evident from Figure 8 that, except near the quiet threshold, the masked threshold goes up 10 db when the level of the noise is raised 10 db. Figure 10 shows the relation between masking (M) in decibels and the effective level (Z) of the masking noise. The effective level Z is the difference in decibels between the pure-tone threshold in quiet and the total energy level in one critical band. In other words, Z is equal to the sensation level of a pure tone having the same energy as the critical band of frequencies which is just able

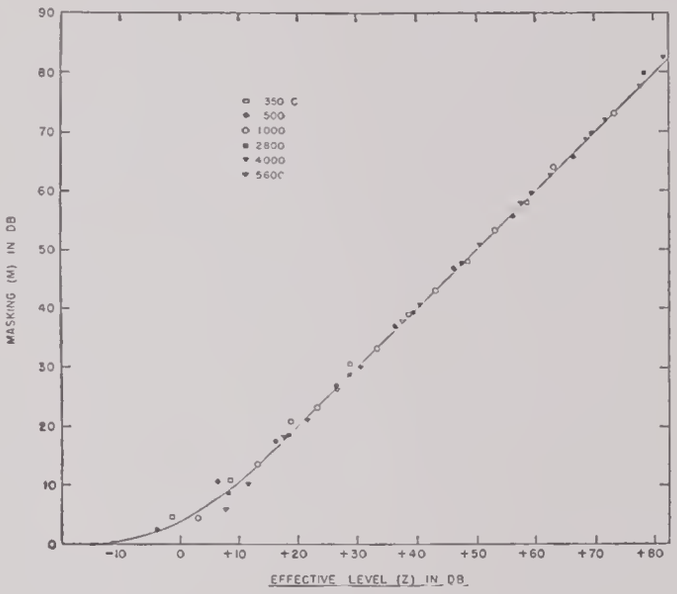


FIGURE 10. Relation between masking (M) and the effective level (Z) of the masking noise.

to mask the tone. Observations at the Psycho-Acoustic Laboratory indicate that Z may be interpreted simply as the sensation level of the critical band. The datum points represent the empirical values for the frequencies, 350, 500, 1,000, 2,800, 4,000, and 5,600 c. The curve drawn through these points has a slope of one for all values of Z greater than 10 db.

3.1.2 Effect of Duration on the Masked Threshold of Tones

The investigations discussed in the preceding section considered only tones of relatively long duration. The effect of shortening the duration of the tone has also been studied.[16a] The results indicate that, as the duration of the tone is increased up to one-fifth of a second, the tone becomes more audible by a proportional amount. The energy required for a short tone to be heard is inversely related to its duration. For

durations longer than one-fifth of a second, the increase in audibility is relatively less as the duration is lengthened. Tones of one-half second duration or longer may be regarded for practical purposes as equivalent to tones of infinite duration. The signal-to-noise ratio (where the noise is expressed in terms of the level per cycle) as a function of the tonal duration is plotted in Figure 11 for four pure tones of different frequencies. Since the shape of the function is independent of the level of the masking noise, the points of Figure 11 were derived by taking the average for the four highest noise levels used. Data obtained in the quiet were less reliable, but seemed to show the same relation. The curve as drawn represents the average for the four frequencies and thus may be considered as a generalized relation between duration and threshold for all frequencies and noise levels. The use of Figure 11 in conjunction with Figure 9 permits an approximate prediction of the threshold for tones of any frequency and duration when masked by a relatively intense random noise of known spectrum.

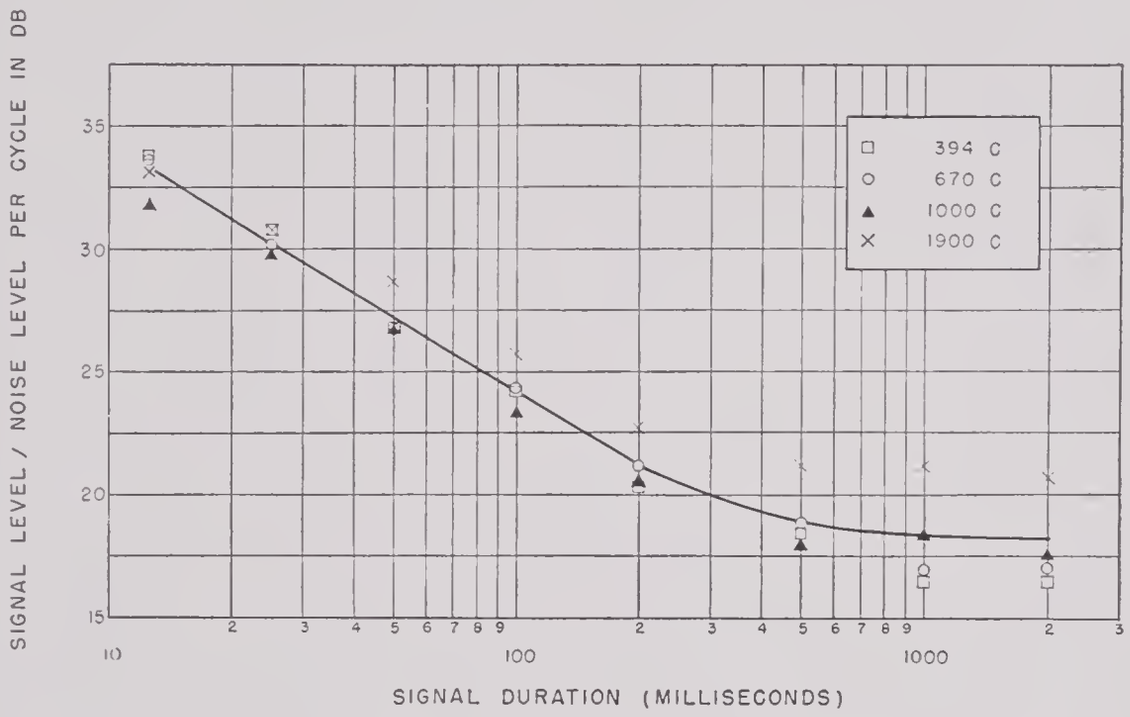


FIGURE 11. The relation between the signal-to-noise ratio (S/N) and the duration of a pure tone at the level of the masked threshold. N is expressed as the noise energy per cycle. Each plotted point is the average S/N for four levels of masking noise (50, 70, 90, and 110 db SPL). Each symbol is for a different frequency as indicated.

3.4.3 Masking of Complex Tones: Applications to Radar

For most military applications, the concern is with the recognition of complex rather than pure tones. The auditory reception of radar echoes, for example, presents the ear with a train of pulses which differ considerably in quality from pure tones. A pulse of short duration is found to contain many harmonics of the fundamental frequency, and if the pulse duration is short enough (50 μsec or less), all the effective audible harmonics are present in equal amounts.

During the war the enemy often detected our radars by auditory means. It was necessary, therefore, to determine what factors contribute to the recognizability of radar pulses and to examine the possibility of disguising the radar in such a way as to make auditory detection more difficult.

The threshold of audibility for pulses of 10 μsec duration was determined for pulse repetition frequencies [PRF] from 1 to 8,000 pulses per second [pps]. A dynamic receiver

(PDR-10) was used, which gave a fairly uniform response up to 8,000 c. For PRF's above approximately 1,000 pps the shape of the threshold function is similar to that obtained with sine waves. From 1,000 down to about 10 pps, however, the root-mean-square [rms] sound-pressure level at threshold remains approximately constant. For a PRF slower than about 6 pps, the individual pulses are heard as isolated clicks, and the threshold depends only upon the peak amplitude of a single pulse. But for the frequencies where discrete pulses are not heard, the ear apparently responds to the higher harmonics which fall in the frequency region to which it is more sensitive. Thus the shape of the threshold function for a complex tone may differ considerably at low frequencies from that for pure tones.

The recognition of a radar signal depends primarily upon three psychological factors: the audibility, the tonality, and the regularity of the incoming signal.[14] Regularity depends upon the sweep rate of the radar and may be modified by varying the sweep. Audibility and tonality are more complex. The operator listening for radar signals must contend with noise, and the ease of detection is dependent to a great extent on the noise level at the time.

The presence of noise suggests that there might be some advantage in randomizing the pulse rate. Methods for randomly triggering a radar set have been investigated in the Psycho-Acoustic Laboratory. A random PRF loses tonality, blends into the background noise, and is more easily masked. Laboratory tests showed that random modulation of the interval between pulses could reduce the audibility as much as 4 db. The tonality of the signal then changes markedly from the buzz of pulses to complete noisiness. When the PRF is modulated by bands of noise containing frequencies from 20 c to one-half the PRF, it is found that all ranges of modulation beyond 50 per cent are judged the same in tonality and that greater ranges of modulation than 50 per cent do not increase the noisiness.

Although the laboratory tests were promising and the results of considerable interest for predicting the audibility of complex tones in noise, field tests conducted at the U. S. Submarine Base, New London, Connecticut, indicated that even the modulated signal is immediately recognized as a radar whenever the signal is strong enough to saturate the receiver. Since, for the receiver tested, there was only a 6-db range between saturation and inaudibility of the steady signal, there was too small a range over which the random pulsing could have an effect.

3.5 DIFFERENTIAL SENSITIVITY

Differential sensitivity refers to the change in the stimulus which is necessary in order to be judged by an observer as just noticeably different. Thus, for example, the smallest difference in frequency which can be detected by ear defines the differential threshold for frequency.

Differential sensitivity to frequency is a function of both frequency and intensity.[4] Above 500 c the *relative* differential thresholds ($\Delta F/F$) are approximately constant, but below 500 c the absolute thresholds (ΔF) are nearly constant. For all frequencies, the maximum sensitivity is obtained at sensation levels of 30 db or more. The value $\Delta F/F$ is smallest for tones of about 2,000 c at sensation levels of 30 to 80 db.

The differential threshold for intensity likewise depends upon the frequency and intensity of the tones.[2] At sensation levels below 20 db, sensitivity varies considerably as a function of frequency, with maximum sensitivity in the region of 2,500 c. The number of decibels required to perceive an increment in loudness decreases for the higher sensation levels, and differences as small as one-fifth of a decibel are detectable under optimal conditions.

The relative differential sensitivity of the ear is greatest for both frequency and intensity over the frequency range which corresponds to the region of maximum absolute sensitivity.

3.5.1 Differential Sensitivity to Noise[15]

Under some circumstances it is possible to harness the noises of mechanized warfare to beneficial effect. In certain tactical maneuvers there is considerable advantage to be gained

through knowledge of the sounds made by vehicles in motion. Cases in point might be: estimation of changes in engine noise as indicating size, direction, speed of approach, or retreat of plane formations; interpretation of noise variations picked up by submarine detection apparatus; relation of changes in air turbulence and propeller noises to changes in speed and altitude of an airplane. Such instances encouraged the development of two phonographically recorded listening tests: a pitch test (Auditory Test No. 6) which measures the listener's ability to detect frequency variations in a noise spectrum, and a loudness test (Auditory Test No. 7) for intensity variations.

The pitch of a band of frequencies is rather indefinitely localized, but appears to be somewhere in the middle range of the frequencies present. To construct a test of differential sensitivity to the frequencies of a noise, it is necessary therefore to make the apparent center move up and down by small amounts. Such changes must be made without an accompanying change in intensity. To accomplish this, the circuit shown schematically in Figure 12 was used.

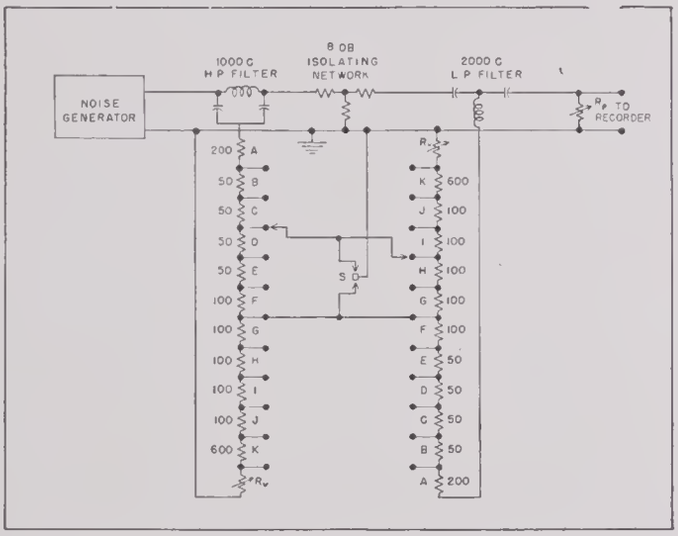


FIGURE 12. Schematic circuit used to record the pitch test.

Each item of the pitch test calls for a comparison of the pitch of two noises. For each item, therefore, the first of the two noises is the "standard," and remains the same throughout the test. The second noise, the "variable," may be higher or lower in pitch than the standard. The center frequency of the standard noise was approximately 1,300 c and had the spectrum shown in Figure 13. Since the center frequency of the bands is difficult to assign

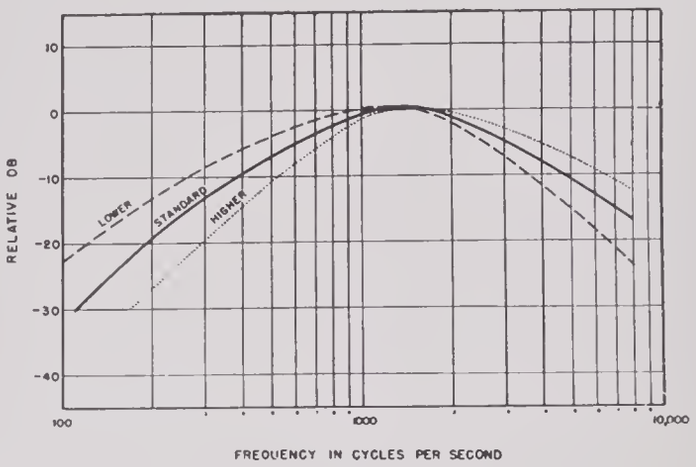


FIGURE 13. Showing the shape of the noise spectra employed for the test of pitch discrimination. The result of shifting the spectrum up or down the frequency scale is shown by the curves marked "Lower" and "Higher."

precisely, it is convenient to pick some arbitrary frequency at a distance from the center frequency and to measure the frequency change in terms of the number of decibels the variable spectrum is above or below the standard at that frequency. If 350 c is chosen as the frequency at which to compare the standard and variable spectrum levels, differences of ± 4 db were obtained for the spectra used in the test. The results obtained with a group of 50 listeners can then be plotted in the form shown in Figure 14. Apparently the ear can detect an astonishingly small shift in the frequency distribution of a complex noise.

One group of 95 listeners took both this test and the Seashore Pitch Test, Form A, which uses pure tones instead of noises. The correlation between the scores on the two tests was 0.41.

A slightly broader band of noise frequencies (500 to 2,000 c) was used for the loudness test. The results for 50 subjects are shown in Figure 15. It will be seen from this figure that judgments of "louder" were easier to make than judgments of "softer," and that the point of subjective equality (50 per cent correct judgments) does not coincide with the point of objective equality (0-db change from standard). This systematic tendency to overestimate the loudness of the second of two stimuli has been attributed to a "time error" which may be

involved in successive comparisons. A similar time error was observed for the Seashore Loudness Test, Form A, which uses pure tones. The correlation between the two tests was 0.39 for a group of 99 listeners.

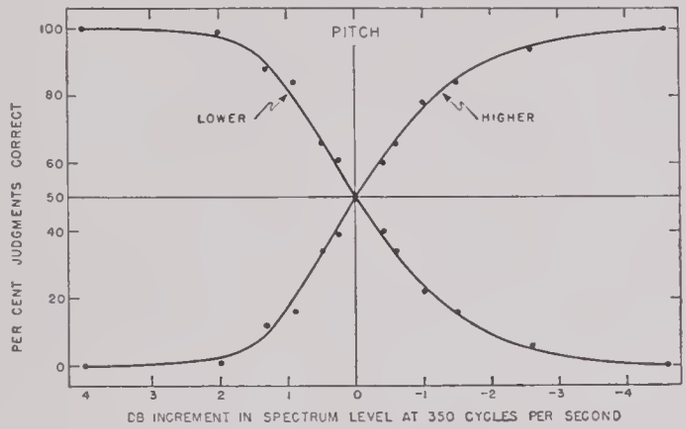


FIGURE 14. Results obtained with pitch test for a group of 50 listeners.

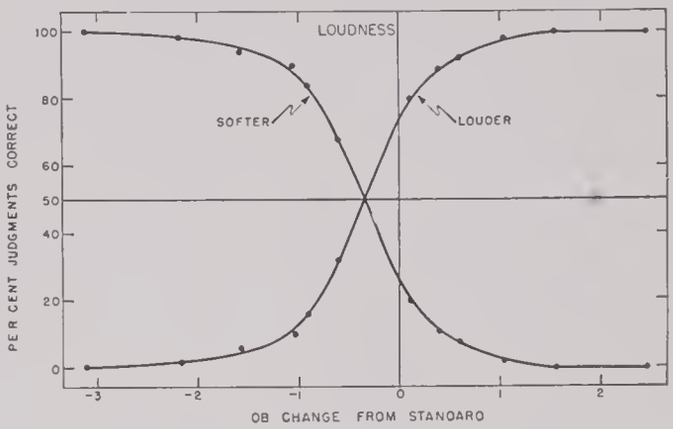


FIGURE 15. Results obtained with loudness test for a group of 50 listeners.

3.5.2 Application to Radio-Range Signals

Radio-range signals are commonly produced by using two directional radio beams with crossed axes. The indicated range is the line along which the field strength of the two beams is equal. Signals are transmitted alternately in some sort of interlocked pattern, and the aviator following the beam can recognize which signal is stronger. The most commonly used signals consist of the Morse Code letters *A* and *N*, interlocked in such a way that the spaces in one are filled by the marks of the other. When aural indication is used, a steady tone results when the plane is flying "on the beam."

It can be seen, therefore, that the pilot's ability to fly along a radio beam will be limited by his ability to discriminate between the *A* and *N* signals. Consequently, the width of the range will depend, in part at least, upon the pilot's differential sensitivity to intensity.

Now, unfortunately, the definitive data on differential sensitivity to intensity cannot be directly applied here, because, whereas the laboratory data were obtained under quiet conditions, the pilot is forced to read the signal against a background of noise. It was necessary, therefore, to determine experimentally the differential threshold at various signal-to-noise ratios.[18]

A heavy, audio "static" was used to provide a continuous, crackling noise. The intensity level of the pure tone (1,000 c) was varied between 86 and 124 db, and at each level the signal-to-noise ratio ranged from −10 to +50 db. The experimental results obtained with a group of 10 listeners are presented in Table 1. The threshold is given in terms of the number of decibels necessary in order to hear the *A* signal 50 per cent of the time.

TABLE 1. Threshold for *A* signal as a function of intensity and signal-to-noise ratio.

S/*N* ratio in decibels	Intensity level of tone in decibels					
	86	96	106	116	121	124
−10	2.30	2.23	2.36	2.81	...	...
− 5	1.46	1.41	1.43	1.57	...	...
0	1.18	1.07	1.10	1.21	1.30	1.40
5	0.86	0.76	0.84	0.97	1.05	1.11
10	0.72	0.62	0.64	0.74	0.80	0.86
20	0.59	0.49	0.46	0.57	0.65	0.70
30	0.58	0.43	0.38	0.41	0.49	0.55
50	0.47	0.38	0.32	0.34	0.36	0.40

It can be seen from inspection of these values that maximum sensitivity at all signal-to-noise ratios occurs with a signal intensity of roughly 96 to 106 db. The effect of adverse noise conditions is a radical decrease in differential sensitivity. From this fact it can be predicted that at extreme distances from the transmitter, where the signal is weak, the radio range may have four to five times the angular width which it has near the transmitter. Several devices which were developed to minimize some of the difficulties in receiving radio-range signals are discussed in Chapter 13.

Chapter 4

SOME CHARACTERISTICS OF HUMAN SPEECH

ANY INVESTIGATION of a voice communication system must take into account the properties of the speech signal which the system is designed to transmit. In this chapter is presented a brief discussion of the important acoustical characteristics of speech, together with a discussion of certain psychophysical problems relating to the recognition of speech sounds.

4.1 SOME ACOUSTICAL CHARACTERISTICS

The production of human speech can be looked upon physically as a process of modulation.[6] A "carrier," whose energy is supplied principally by the vocal cords, driven by the air stream from the talker's lungs, is "modulated" by the speech message. This modulation is accomplished by appropriate movements and adjustments of the vocal equipment of the talker, notably the tongue, the lips, the teeth, and the cavities of the mouth, nose, and throat. These movements take place at the phonemic rate (about 10 per sec). The carrier for the voiced sounds is produced by vibration of the vocal folds. The unvoiced sounds are generated by breath noise resulting from forcing a stream of air past suitable constrictions in the vocal passage. In general, the frequency spectra of vowels consist of discrete frequency components with fundamentals in the neighborhood of 100 to 200 c. The consonants, however, contain high-frequency components having a nearly continuous distribution of energy.

4.1.1 The Speech Spectrum

The frequency spectrum of connected discourse contains not only the frequency components of the complex carriers but also the frequency components corresponding to the modulation. This includes the starting and stopping of sounds, changing of carriers, and changing of pitch. Hence, the "instantaneous" spectrum undergoes continuous change. The term spectrum which is borrowed from optics usually applies to steady-state phenomena. Its use in describing speech is justified however, if the contributions of the speech sounds for the various frequencies are suitably averaged over periods equal to the duration of a syllable or longer. The statistics of instantaneous amplitude vs frequency distribution is closely linked with the ever changing instantaneous spectrum. The amplitude distribution of conversational speech is taken up in Section 4.1.2.

The speech spectrum is best investigated experimentally by measuring the spectrum of the free-field sound pressure when the talker is located in a room free from acoustic wall reflections (anechoic chamber). The exploring microphone is usually inserted into the sound field produced by the talker at a certain distance directly in front of the talker's lips. For positions along such a line extending perpendicularly away from the talker's lips, the spectrum of speech is within wide limits independent of the distance from the lips and it is appropriate to apply the inverse square law relating intensity and distance.[5] The speech spectrum is then determined by analyzing the output voltage of the exploring microphone by means of a series of band-pass filters with contiguous pass bands. The output of each filter and the total, or overall, output are measured by a suitable instrument possessing a square-law characteristic and averaged over a period of at least several seconds. A thermocouple and fluxmeter[4] was used in studies at the Bell Telephone Laboratories. At the Electro-Acoustic Laboratory a vacuum-tube circuit with parabolic characteristics was employed.

An analyzing device of this type can appropriately be called an audio-spectrometer. It is convenient to express the output in each band in terms of root-mean-square [rms] sound pressure per cycle in decibels relative to the standard reference level 0.0002 dyne/cm^2 which is called the spectrum level [SL].[3]

Figure 1 shows the average spectra measured

at the Bell Telephone Laboratories[7] for a crew of six men and five women. The results are expressed in terms of the free-field rms sound-pressure levels per cycle [SL] 30 cm directly in front of the talkers' lips. Circles are drawn at the mid-band frequencies of the filters used. The speech material was obtained by reading aloud from English stories at what was considered to be conversational speech levels.

Figure 2 shows the average spectrum for a crew of seven young men as measured with the audio-spectrometer at the Electro-Acoustic Laboratory.[10] A photograph of the equipment is shown in Figure 3. The speech material used

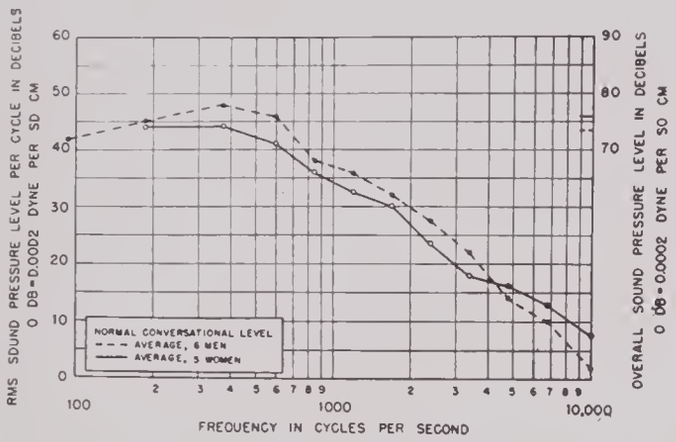


FIGURE 1. Average speech spectra at a distance of 30 cm from the talkers' lips. (BTL)

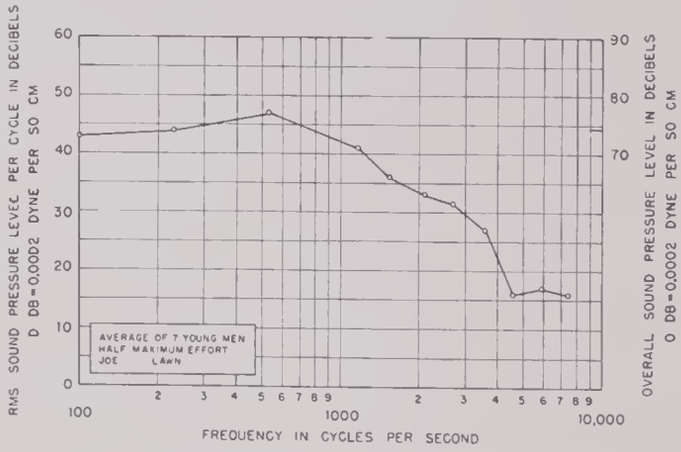
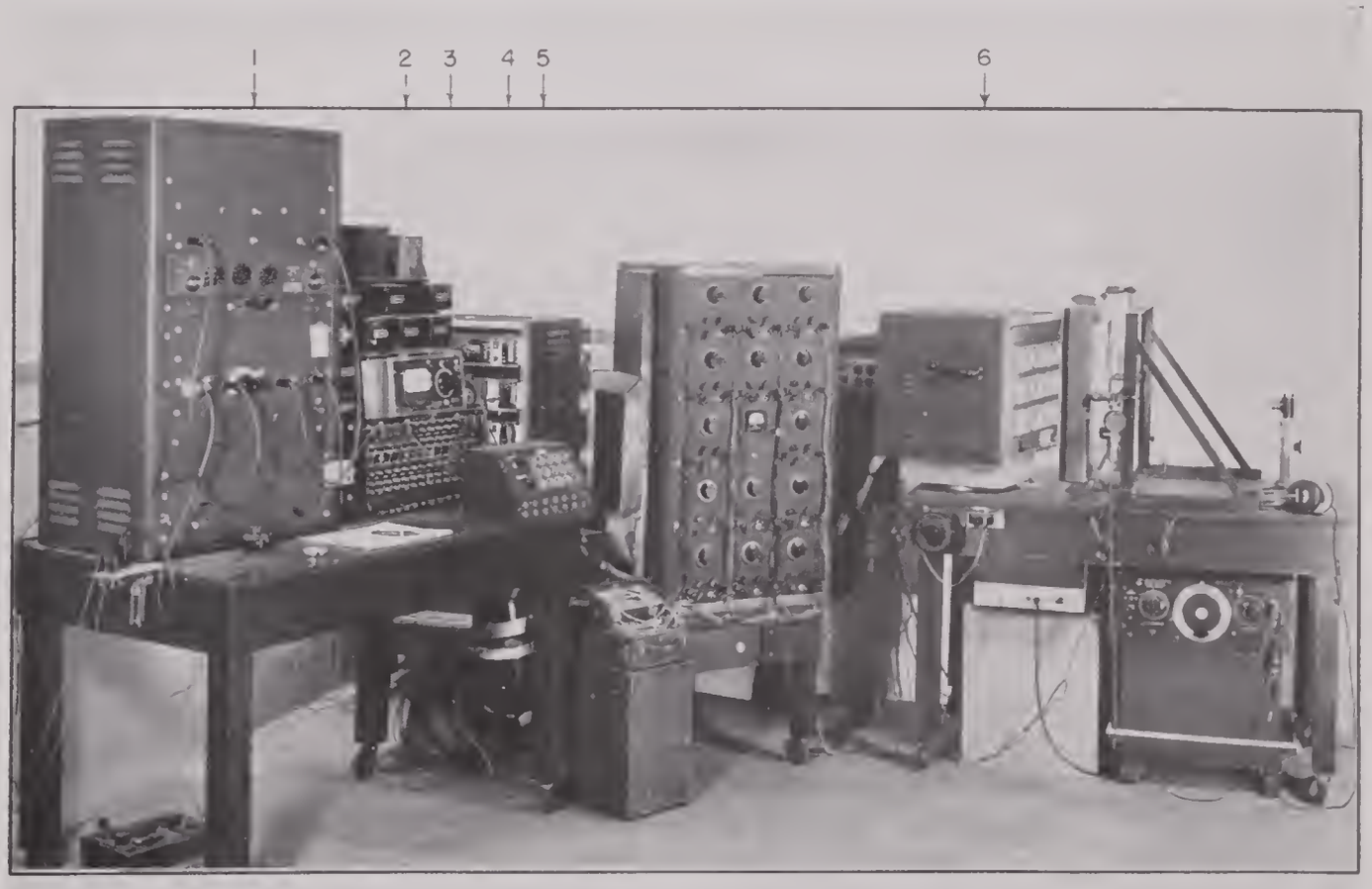


FIGURE 2. Average speech spectrum at a distance of 18 in. from the talkers' lips (Harvard)



1. INPUT AMPLIFIER
2. BAND-PASS FILTERS
3. VU MONITORING METER
4. POWER SUPPLIES
5. NEON OVERLOAD INDICATORS
6. SQUARE-LAW INTEGRATORS

FIGURE 3. The audio-spectrometer.

was the repeated test sentence: "Joe took father's shoebench out; she was waiting at my lawn." This sentence, devised by the Bell Telephone Laboratories, is said to represent a cross section of the most important speech sounds

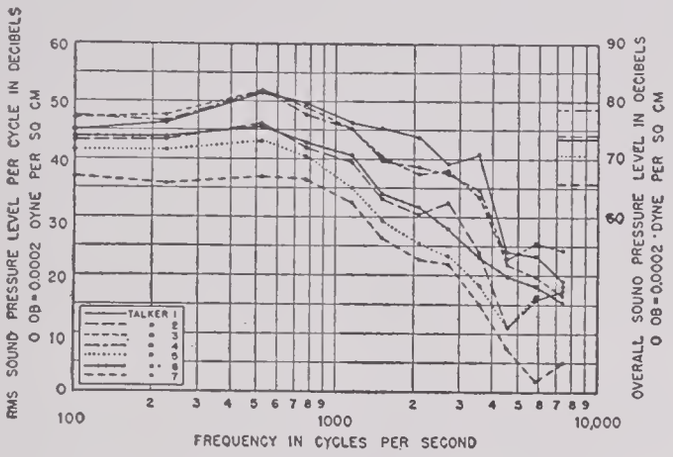


FIGURE 4. Variation of the speech spectrum with different talkers.

used in conversational English. The speaking level of the talker was maintained constant at 6 db below what each talker decided to be his level of "maximum (sustained) effort," (see Section 4.1.3). The monitoring was accomplished by reading the frequent peaks in the test sentence on a standard volume indicator (VU meter).[a]

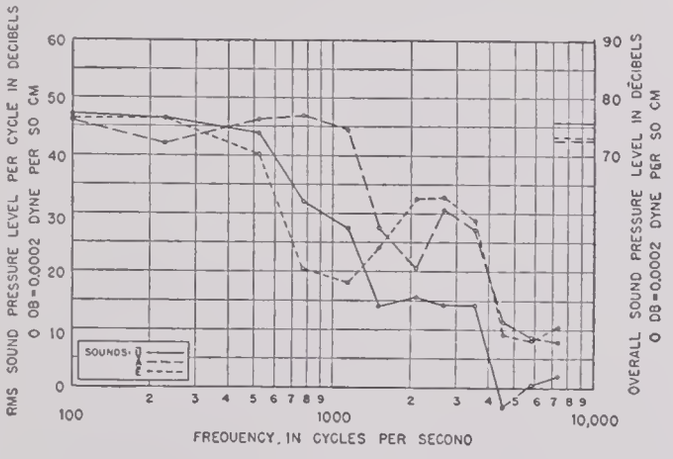


FIGURE 5. Average spectra for three vowels.

After allowing for the different distances of the microphone from the talkers' lips and considering the different speaking levels, the agreement between Figures 1 and 2 is reasonably good, especially since variations from speaker to speaker are considerable (see Figure 4). Variations in a given speaker from one test to the next are of secondary importance[10] as compared with the variations between different speakers.

It is also desirable to know the spectra of individual speech sounds. The spoken sounds of English can be classified in several ways. The vowels can be arranged in the so-called "vowel triangle" on the basis of the position of the tongue and lips when forming the different sounds.[11b] Figure 5 shows the long-interval rms sound pressures for the vowels ū, ā, ē, averaged for seven men. The spectra for the vowels ō, á, and i, are shown in Figure 6. Figure 7 shows the spectrum of the unvoiced fricative consonant sh with its predominant high-frequency energy.

Although the high-frequency energy content of certain consonants is considerable, they tend to be of short duration in comparison with vowels, and they contribute comparatively little to the total energy in connected discourse. Although consonants are of prime importance for the intelligibility of speech, physical measurements of the type discussed do not always provide a suitable tool for determining the

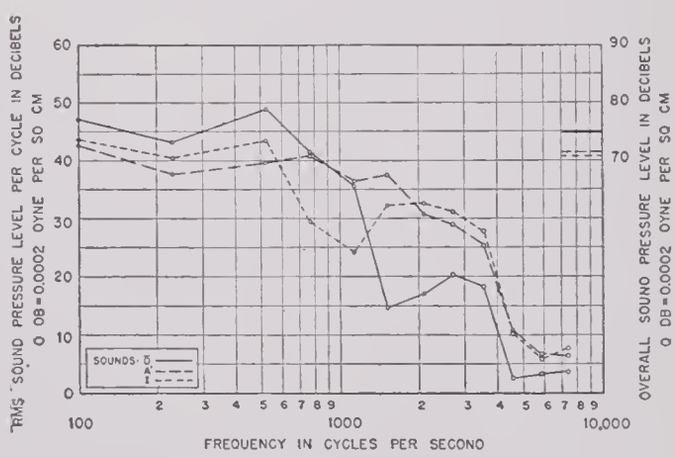


FIGURE 6. Average spectra for three vowels.

presence or absence of consonants if they occur in combination with vowels. Figure 8 provides a typical example. The spectra of two words of the consonant-vowel-consonant type are compared with the spectrum of the vowel (ē) alone. The difference between the three spectra is insignificant.

These facts make it clear that an audio-

[a] Measurements have shown that the average of the deflections determined on the frequent peaks (for unfiltered speech) of a standard VU meter is about 3 db higher than the long-interval rms value of speech.

spectrometer of the type described above cannot be used for direct evaluation of the intelligence conveyed by speech.[b] Intelligibility (articulation) tests have to be resorted to for this purpose. Nevertheless, information of the type discussed in this section is valuable in providing some engineering information relative to the design of voice communication systems and their components.

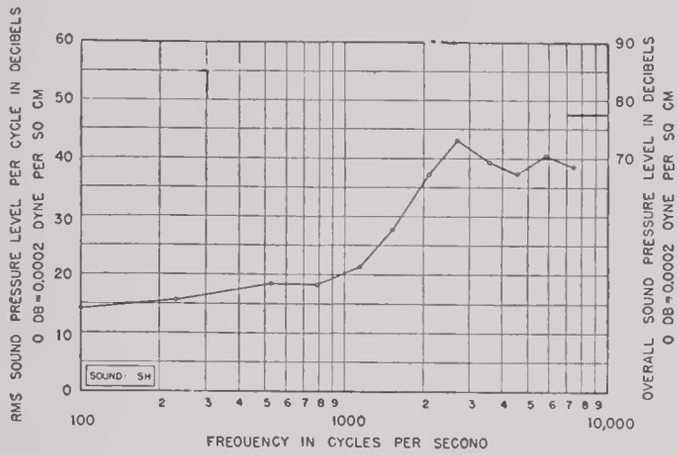


FIGURE 7. Average spectrum for the sound "sh."

any given instant, however, the levels fluctuate widely about these average values. The sound-pressure distribution as a function of time and frequency is of importance for problems concerning the intelligibility of speech and for proper design of voice communication systems. This distribution of speech levels was studied at the Bell Telephone Laboratories.[7a] By means of a suitable audio-spectrometer and an arrangement of thyratron circuits and relays, the root-mean-square and peak sound pressures in intervals of ⅛ sec (approximate duration of a phoneme) were measured as a function of time and frequency. Figure 9 shows a plot of the *peak* sound pressures in ⅛-sec intervals for a crew of six men as a function of frequency. The parameter is the percentage of intervals in which the pressure exceeded

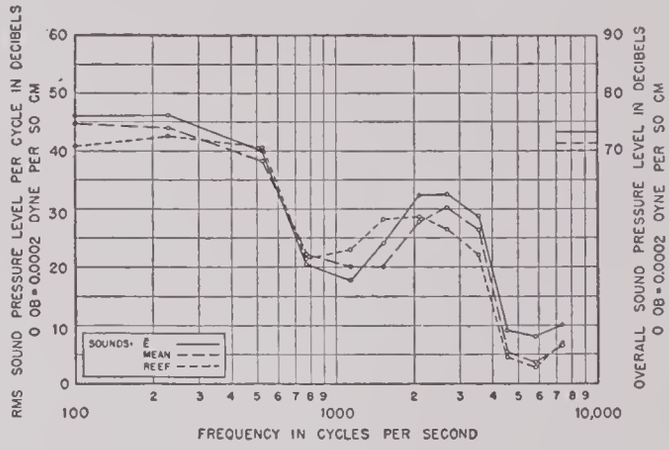


FIGURE 8. Comparison of the average spectra of two words and the vowel which they contain.

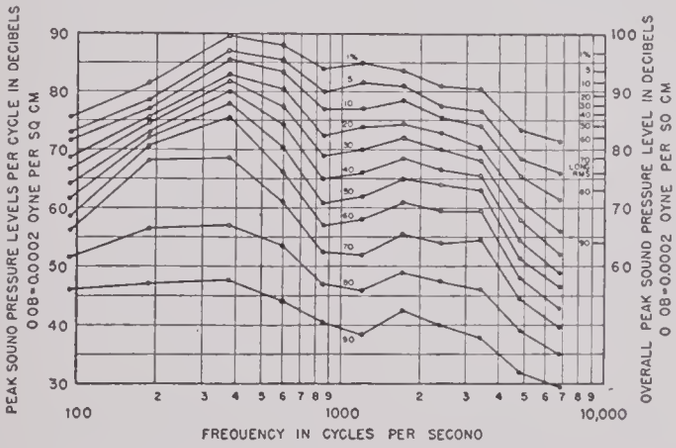


FIGURE 9. Peak pressures in ⅛-sec intervals for conversational speech at a distance of 30 cm from the talkers' lips. (Average of 6 men.) (BTL)

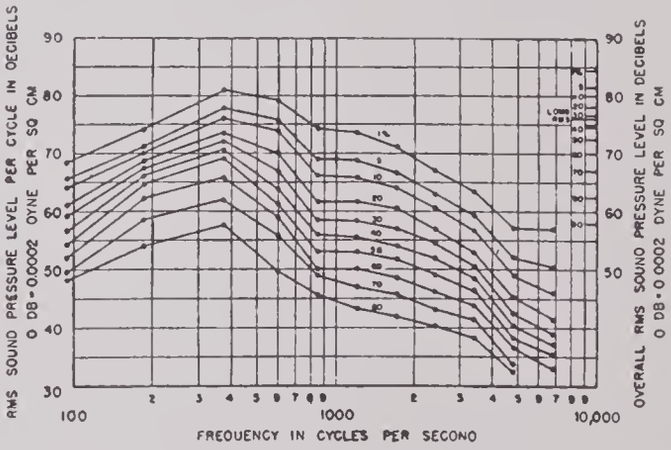


FIGURE 10. RMS pressures in ⅛-sec intervals for conversational speech at a distance of 30 cm from the talkers' lips. (Average of 6 men.) (BTL)

4.1.2 The Amplitude Distribution of Speech

The spectra of speech which have been discussed in the preceding section represent the long-interval rms sound-pressure levels averaged over an appreciable period of time. At

[b] Progress in this direction has been made at the Bell Telephone Laboratories, using a special form of visual presentation of the analyzed speech.[11]

the value plotted as ordinate. For completeness, the level distribution for unfiltered speech is also given.

Figure 10 shows a similar plot of the rms value in ⅛-sec intervals. The long-interval (average) rms sound pressures for all ⅛-sec intervals provide a basis for drawing a cumulative distribution curve. Figure 11 shows such

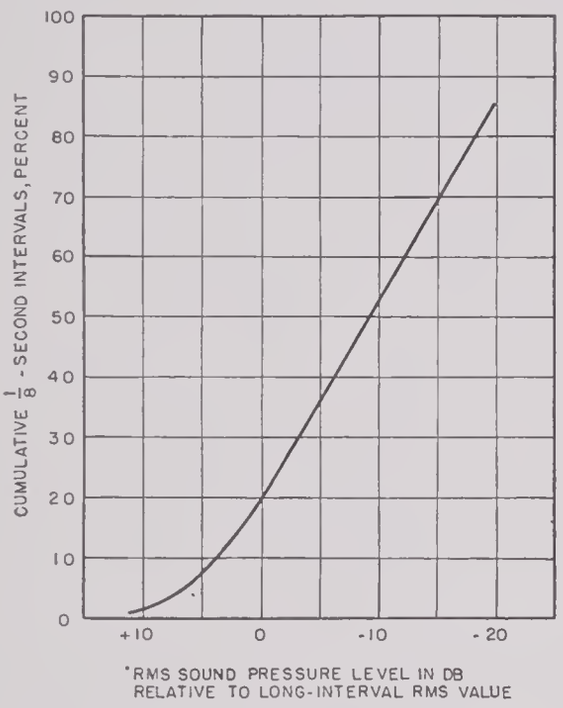


FIGURE 11. Cumulative level distribution of conversational speech. (BTL)

a curve for the band 1,000 to 1,400 c. The ordinate shows the percentage of ⅛-sec intervals in which the rms sound pressure exceeded the value given by the abscissa in terms of decibels above or below the long-average rms sound pressure. From the fact that the contours in Figure 10 are approximately parallel, it can be concluded that Figure 11 is approximately applicable over the whole frequency range.

It has been shown[7a] that the results are not greatly altered if intervals of ¼-sec duration are used.

If the assumption is made that intervals arranged in order of increasing peak pressures would show the same order when rms pressures are considered, then the peak factor (ratio of peak to rms pressure, in decibels) can be computed for any given contour in Figures 9 and 10 and can be taken to be the peak factor of speech in those ⅛-sec intervals. In Figure 12 the peak factor for the 10 per cent contour is shown as a function of frequency and also for unfiltered speech. Naturally, the peak factor for long intervals is also of interest and is shown in Figure 13. It means that, in unfiltered English speech, peaks occur about 20 db above the long-average rms value if the interval of observation is of the order of one minute. Clearly, information of this type is important for the design of communication equipment carrying voice signals.

1.1.3 Speech at Altitude

In the early years of World War II numerous complaints from aviators on high-altitude missions reported a significant deterioration in the quality of voice signals. This problem was judged to be of such importance that in 1942 a research program was undertaken at the Electro-Acoustic Laboratory to study quantitatively the effects of reduced ambient pressure on human talkers and listeners. A large number of controlled experiments were made using a crew of trained men in an altitude chamber.[10] Simultaneously, controlled articulation tests were carried out by another group of investi-

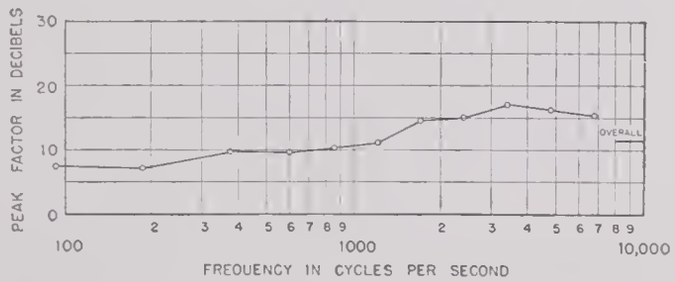


FIGURE 12. Peak factor of conversational speech at the 10 per cent level of ⅛-sec intervals. (BTL)

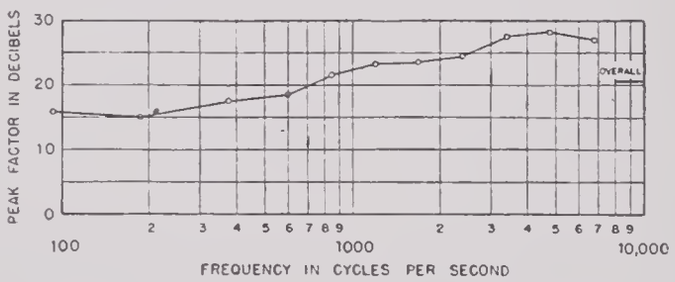


FIGURE 13. Peak factor of conversational speech for long intervals.

gators from the Psycho-Acoustic and Electro-Acoustic Laboratories in high-flying airplanes under conditions much like those encountered on long-range bombing missions. This latter phase of the research is discussed in Chapter 9.

In this section a brief discussion is given of some of the physical parameters of speech affected by reduced ambient pressure. The effects of altitude on the human voice alone is considered. The behavior of the voice at altitude when an oxygen mask is worn is discussed in Chapter 9.

To determine to what extent the change in the talker's voice is alone responsible for the altitude decrement observed in the field, data were taken at a simulated altitude of 35,000 ft.[c] The subjects removed their oxygen masks temporarily and spoke into a miniature condenser microphone 18 in. directly in front of their lips. The output voltage of the microphone was analyzed in frequency bands by means of the audio-spectrometer. The same procedure was repeated at sea level. The ratio in decibels of the sound pressures at sea level and at altitude, the altitude decrement [AD], is plotted

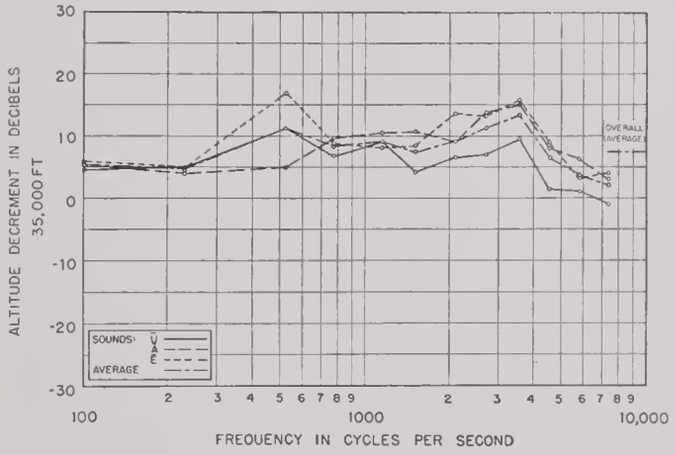


FIGURE 14. Average altitude decrement of the voice alone at constant effort for various vowels.

in Figure 14 as a function of frequency. The vowels ū, ā, ē, were used in this study. The AD is not greatly different for the three vowels chosen but is positive in all three cases: the sound pressures at altitude are smaller than those at sea level. Since the five speakers used in these experiments experienced certain difficulties in enunciation and timing due to the temporarily interrupted supply of oxygen, the results of Figure 14 can be regarded only as approximations.

For reasons explained in Section 4.1.1 the altitude decrements of most consonants cannot be evaluated directly by an audio-spectrometer of conventional design. To accomplish this, photographs of the trace of the beam on an oscilloscope were made at sea level and 35,000 ft altitude.[10] The standard test sentence, "Joe took father's shoebench out; she is waiting at my lawn," was used. Again the mask was removed during the process of speaking. Sample records are given for two speakers in Figures 15 and 16. The traces represent replicas of the output voltage of the condenser microphone located in front of the talker's lips. Since, in going from sea level to altitude, suitable adjustments in the overall amplifier gain had to be made, the records are best evaluated by comparing the ratio of the deflections for a given consonant with the deflections of the adjacent vowel or vowels. Figure 17 shows the average *gain* in level for ten consonants and semivowels with respect to the adjacent vowels in the test sentence. The bars represent the averages for four talkers. The fricative and stop consonants (j, k, f, sh, ch, t) show a gain of from 10 to 15 db. The semivowels (l, m, n, ng) show no appreciable net gain. The gain in level of the fricative and stop consonants more than offsets the average altitude decrement of the vowels (see Figure 14), and as a result consonants sound somewhat more distinct in speech at altitude. Offsetting this effect, however, is a "booming" phenomenon at low frequencies when oxygen masks are used. Most of those who spoke into open air at altitude noticed some difficulty in effecting nasal resonances and described feelings of a "stuffy nose" or "cottony throat."

In all speech measurements of the type discussed, the question of "monitoring" the voice is important. In experiments at sea level suitable monitoring microphones operating a volume indicator visible to the talker can be used to keep the talker's voice "constant" from

[c] By altitude is meant the so-called pressure altitude. In the U. S. Standard Atmosphere, for example, an altitude of 35,000 ft corresponds to an ambient pressure of 177 mm Hg. For a discussion of pressure vs density altitude, see Report OSRD 3106.[10]

test to test. This is not the case with measurements at altitude. It is pointless to attempt to speak at the same level at all altitudes if the very changes in voice quality and levels are themselves to be studied. A "normal" speaking voice could not be chosen for similar reasons.

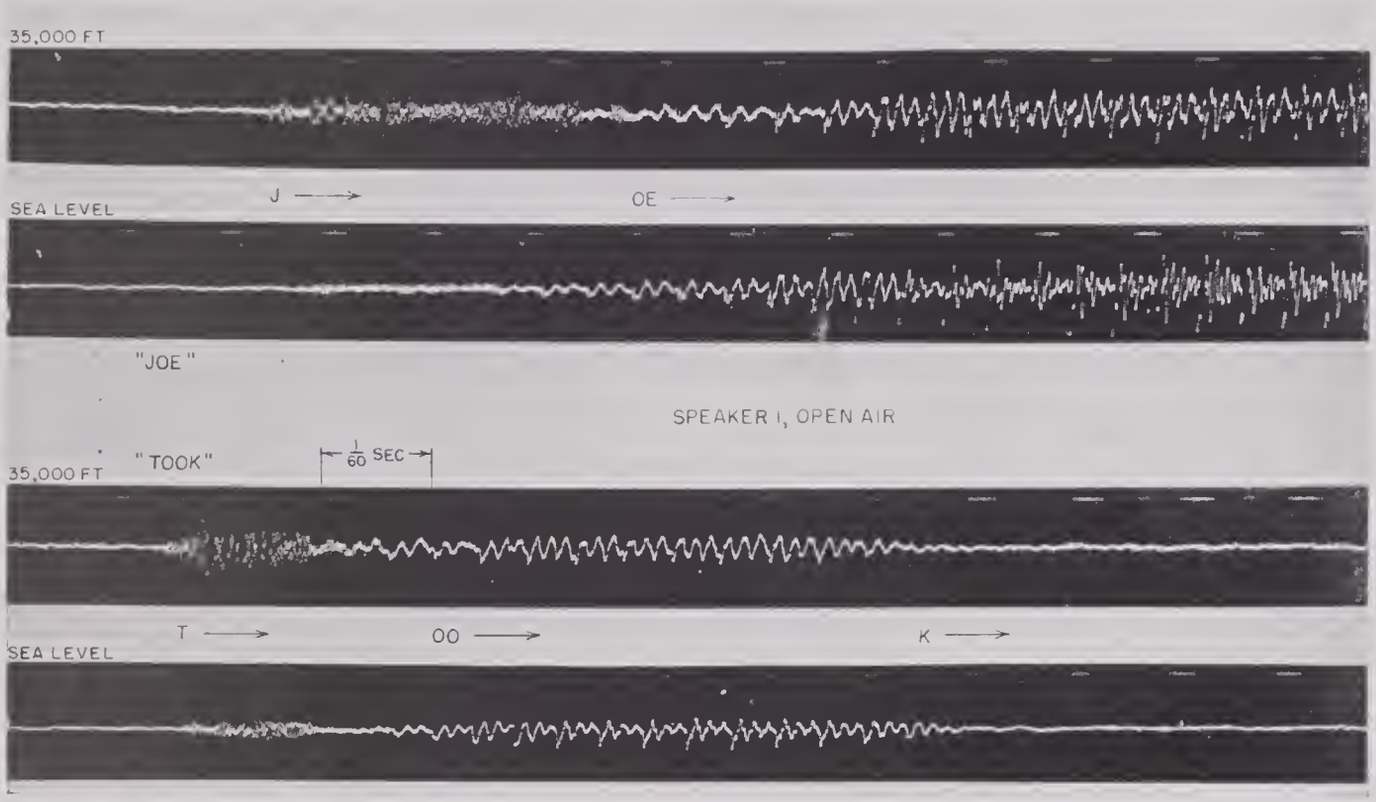


FIGURE 15. High-speed photographs of speech waves at sea level and 35,000 ft.

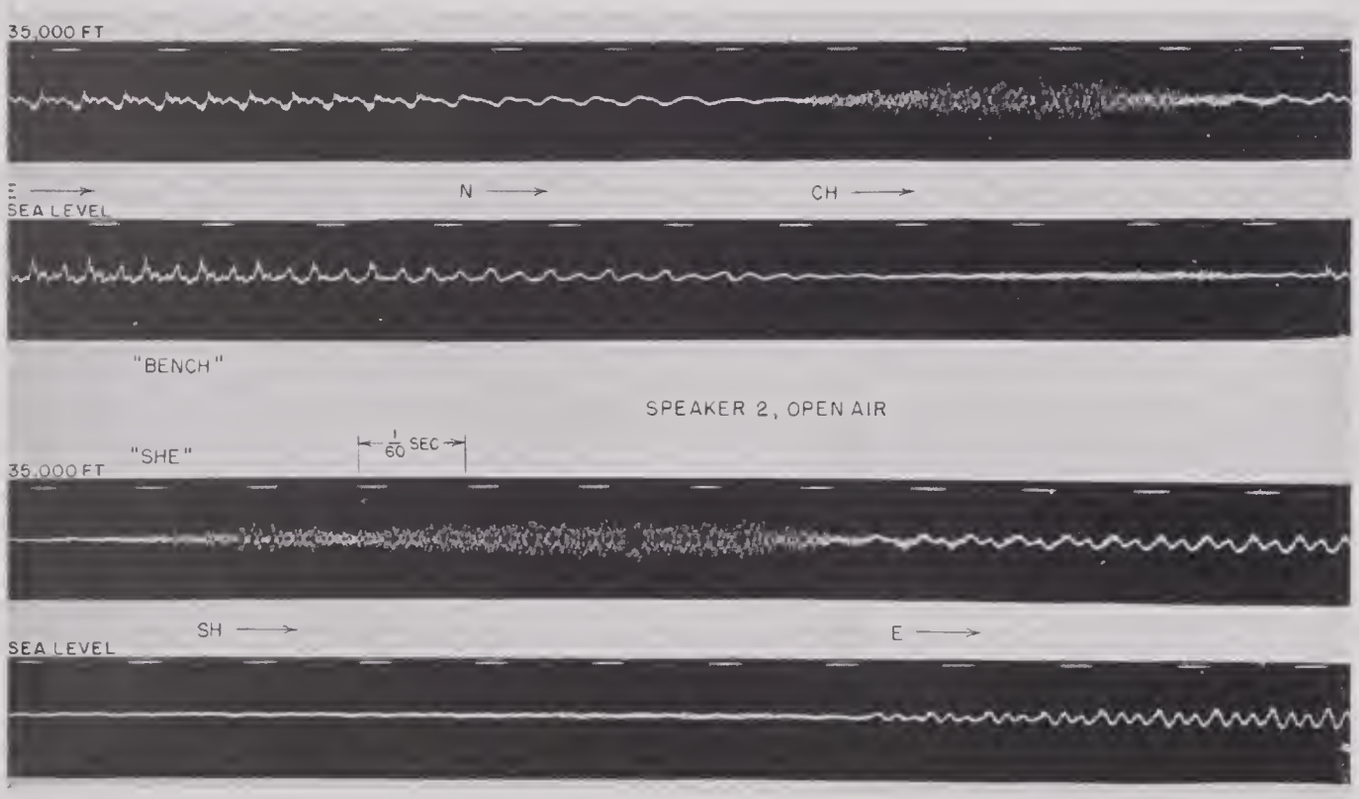


FIGURE 16. High-speed photographs of speech waves at sea level and 35,000 ft.

Side-tone[d] monitoring was deemed undesirable because of its wide variability and dependence on the equipment used.

A duplicable level of talking was found to be that of *half effort*. The subject was first required to pronounce all sounds as loudly as

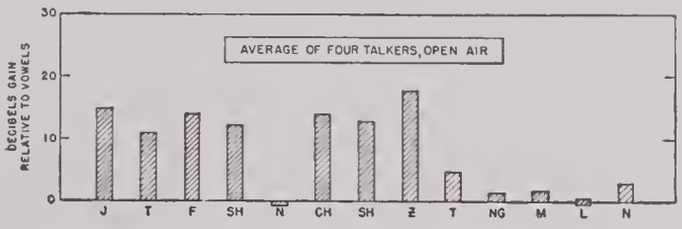


FIGURE 17. Average gain of consonants and semivowels relative to the adjacent vowels at 35,000 ft.

possible. He was then required to talk at what he considered half maximum effort. Long series of measurements confirmed the fact that the talkers make the judgment of half effort consistently, at both sea level and altitude. In terms of volume-indicator readings, the difference between maximum and half effort corresponds to about 6 db, and this monitoring level was used during all the tests described above.

Subjective estimates by listeners during several months of testing at the Electro-Acoustic Laboratory indicate that the *average* pitch at which a person talks does not vary appreciably between sea level and 40,000 ft.

The number of words which can be said on one breath at a constant output level was found to be proportional to the air density. In normal speaking at 35,000 ft it is necessary to pause for breath two or three times as often as at sea level.

1.1.4 Peak Factors of Vowels

In Figure 18 the vowel sounds have been arranged in the "vowel triangle."[1] The size of the peak factor, as measured for a sample of 47 male talkers, is indicated by the linear height of the bar for each vowel.

Since the vowels have been arranged in the triangle on the basis of the position of the tongue in the mouth in forming the vowel, those on the left are back vowels and those on the right are front vowels. Those at the bottom of the triangle are formed with the tongue resting near the floor of the mouth. Front and back vowels are normally spoken with the lips near together, while vowels at the lower corner are formed with the lips apart.

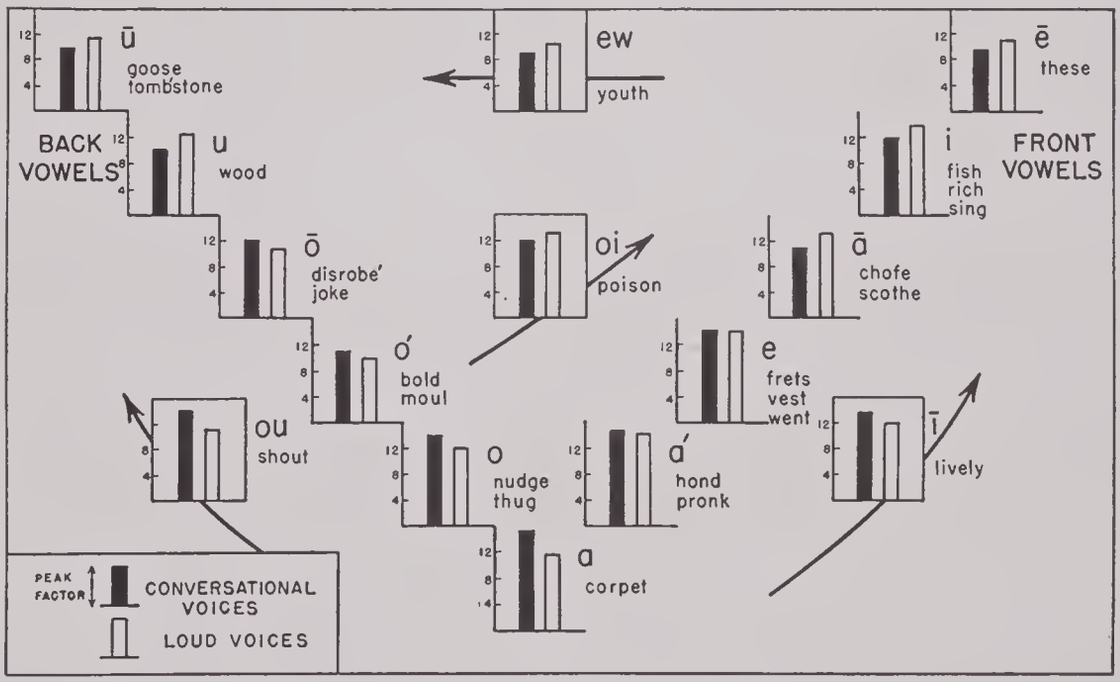


FIGURE 18. Peak factors of vowels for conversational and loud voices.

Examination of the peak factors measured for speech at loud and conversational levels

[d] A method of monitoring which diverts a part of the outgoing speech energy (side-tone) to an earphone worn by the talker.

(shaded bars) reveals a curious effect. When the tongue is near the floor of the mouth and the lips are far apart, the peak factor is greatest. When the announcer is using a very loud voice level, however, the peak factors are smaller.

1.2 FACTORS GOVERNING THE RECOGNITION OF SPEECH SOUNDS

The factors which determine whether or not any particular speech sound will be recognized on any given occasion are many and varied. If, however, we simplify the experimental situation by eliminating, i.e., averaging out, the different abilities and idiosyncrasies of different speakers and listeners[e] and by minimizing the distortion in the transmitting network, we find that some speech sounds are intrinsically more recognizable than others. These differences seem to depend upon both the physical and psychological nature of spoken language.

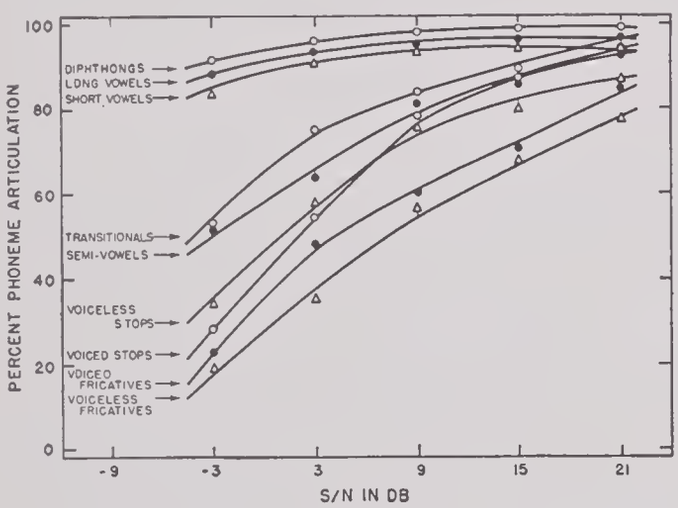


FIGURE 19. Showing the articulation of different classes of speech sounds as a function of the S/N ratio. Nonsense syllables were spoken into a linear system and heard against a background of white-noise interference in the headphones.

Illustrative data are shown in Figure 19, where the percentages of correct identifications of different classes of speech sounds (the classification used here was taken from the literature)[1a] are plotted against the intensity ratio (in decibels) between the speech and a white-noise interference introduced electrically into the earphones. These data were obtained with a system having a relatively uniform frequency response. The test material was composed of nonsense syllables (consonant-vowel and vowel-consonant combinations), and in this way the contextual identification of the sounds was avoided.

Examination of the functions reveals that vowel identification is more accurate than consonant identification. This difference is correlated with the greater speech power of the vowels. It has been estimated, for example, that the most powerful phoneme ("aw" as in *all*) is 680 times (28 db) more intense than the weakest phoneme ("th" as in *thin*).[1c]

Although a small percentage of the incorrect identifications were simple omissions, the usual mistakes (at the S/N ratios used in the experiments) consisted of substituting some other speech sound in place of the sound which was spoken. For convenience, such incorrect identifications can be called "confusions," and it is a matter of some interest to discover which speech sounds are most easily confused.

Analysis shows that the most frequent confusions of consonant sounds are between members of the same class. Thus, voiced stops are most often confused with other voiced stops, unvoiced stops with other unvoiced stops, etc. Less common, but of some importance, were confusions within the voiced sounds or the unvoiced sounds. Confusions of voiced with unvoiced consonants (and vice versa) were relatively rare.

Data illustrating these relations are presented in a graphic table in Figure 20.[f] The sounds spoken are grouped by classes along the top, and the class of the substituted sound is indicated vertically. The shaded area corresponds to the percentage of confusions of one class of sound with another. In Figure 20A the predominance of intraclass confusions is apparent. These data were obtained when the

[e] Individual differences in speakers and listeners, and their effects upon intelligibility are considered in detail in Chapter 14.

[f] These data were selected from a number of analyses made on data collected over a period of 6 months. They can be considered typical of the results obtained with a wide variety of talkers, noises, and equipments when the tests are conducted with meaningless syllables.

interfering signal (in this case, a train of 150 "spikes" per second) was considerably weaker (21 db) than the speech. When the intensity of the interference is increased, however, such confusions become less predominant. Figure 20B shows that, with a speech-to-noise ratio of −3 db, the percentage of confusions between sound classes is greater. Generalizations of this type proved valuable in the construction of special, highly intelligible vocabularies for military use (see Chapter 6).

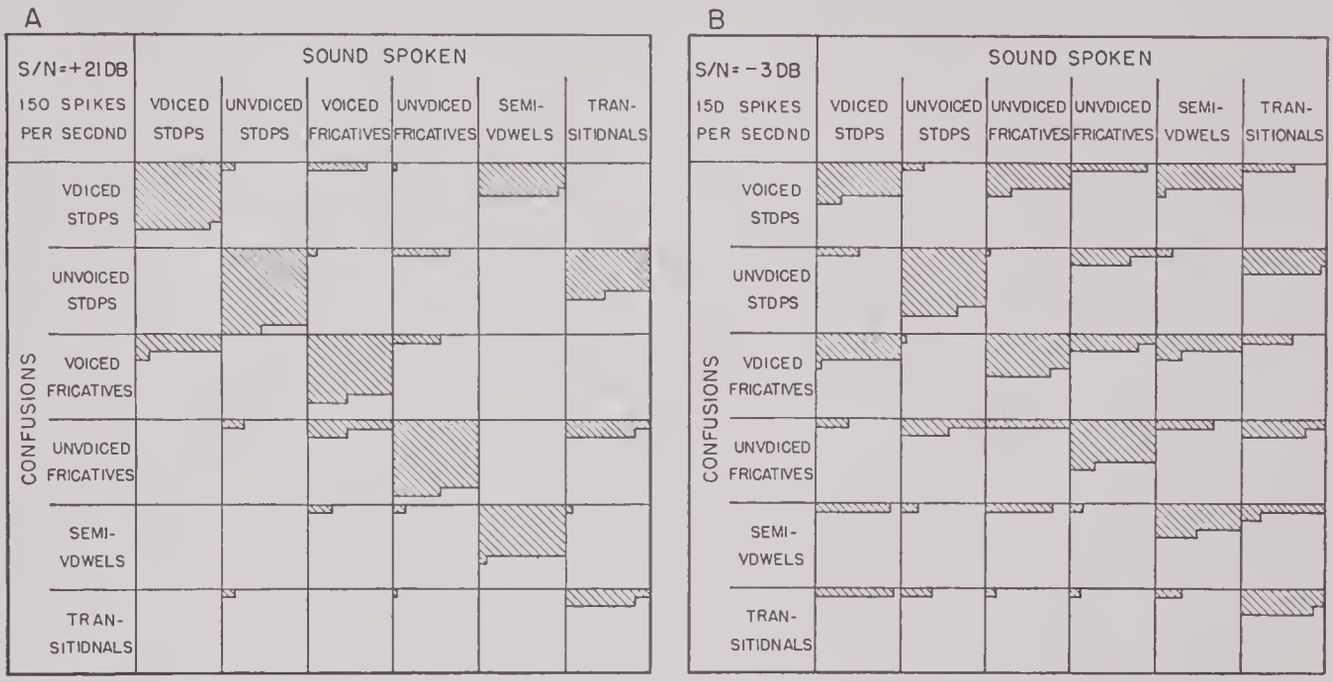


FIGURE 20. Illustrating the types of confusions obtained at low (plot A) and at high (plot B) intensities of the interfering signal. The interference in this instance was provided by a train of 150 spikes per second. At high intensities of the interference, the interclass confusions are increased.

Confusion of one speech sound with another depends primarily upon similarities in the frequency spectra of certain sounds. For each speech sound, certain parts of the characteristic spectrum seem to be the critical regions with respect to identifiability.[14] Consequently, the relation of the spectrum of a particular speech sound to the spectrum of the interfering noise and to the frequency response of the communication system affects its relative identifiability. More attention will be paid to these interactions in later chapters, so it is sufficient to report here that ratings of word identifiability obtained on systems with extreme frequency and amplitude distortion showed significant correlations (0.23 to 0.79) with ratings obtained using noise interference.[9] Thus it is possible to obtain a general list of easily identifiable words, but in order to obtain the maximum effectiveness over any given communication system, the vocabulary should be chosen for that system alone.

Although the spectra and intensities of the speech sounds are of fundamental importance, the intelligibility of speech seems to depend upon other factors than the physical dimensions of the component sounds. Against the roar of noise in a modern battle craft, few words are heard in their entirety. Rather, only certain sounds are recognized, and these must serve as cues from which the listener reconstructs the words spoken into the microphone. Naturally, the more speech sounds in a word, the greater the chance that enough of them will be recognizable for the word to be identified in full. It may also be said that, in general, the longer a word is the smaller is the number of other words sufficiently similar in the overall pattern of their speech sounds to make them mutually subject to confusion. To illustrate this relation, five groups of 20 words, having different average numbers of sounds per word, were read 20 times to a group of listeners. The results of the experiment are shown in

Figure 21. Similar results have been obtained by the Bell Telephone Laboratories.[2]

The readiness with which a word will be identified is clearly a function of its relation to

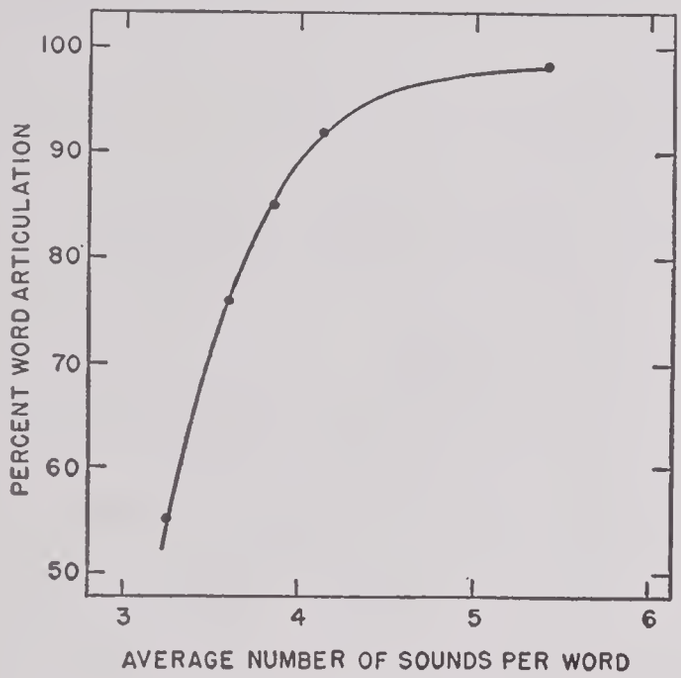


FIGURE 21. Illustrating the improvement in articulation as the average number of speech sounds per word is increased.

other words in the language. For example, the more frequently it is used in everyday discourse, the greater the chances that a mutilated portion of the word will evoke the correct response from the listener. Again, the chance that a word heard only in part will be rightly identified tends to vary inversely with the number of other words it resembles. For example, if the word "power," partially masked, is heard only as "ower," it will often be identified as "tower" or "cower" and sometimes as "bower," "dour," "lour," "hour," "sour," or "flower." In other words, its measured identifiability will be lowered by the fact that many other words happen to differ from "power" only with respect to its initial and relatively low-powered voiceless stop, "p."

Finally, it is necessary to consider as "apperceptive variables" those elements in the experience of the listeners which alter the test results. As apperceptive factors one must include, for example, the relation of the dialect of the listener to the dialect of the speaker. Also important is the educational level of the listener, for if the listener is unfamiliar with the word, he may miss it completely or confuse it with similar words. Related to this is the context, which establishes the listener's expectancy. If the listener knows the stimulus word is a member of a set of words (for example, numerals or phonetic equivalents), then the listener will be prepared for words in that set.

These factors, physical and psychological, appear to be the prime determinants of word identification. It would be advisable to conduct further experiments to evaluate the relative importance of each factor, in which case these generalizations may prove useful as a guide to the design of more basic experimentation.

Chapter 5

ARTICULATION TESTING METHODS

A QUANTITATIVE MEASURE of the intelligibility of speech may be obtained by counting the number of discrete speech units correctly recognized by a listener. The procedure by which this quantitative measure is obtained is an *articulation test*. Typically, an announcer reads lists of syllables, words, or sentences to a group of listeners, and the percentage of items correctly recorded by these listeners is called the *articulation score*. This percentage is taken as a measure of the intelligibility of speech.

Articulation testing methods grew out of the early work of developing the telephone, and the Bell Telephone Laboratories were first to standardize the procedure as a rigorous method of evaluation.[2, 3] Tests of this type have proved useful for comparison of communication devices, evaluation of the effects of noises on communication, measurement of the basic audibility of words and commands, and for rating and training communications personnel.

All articulation scores are relative scores, contingent upon the use of specific announcers, microphones, amplifiers, earphones, noises, listeners, and test lists. Little trust can be placed in absolute statements about articulation, especially if such statements are based on tests with different voices and different listeners. In general, the only trustworthy statements regarding the effectiveness of communication systems are relative statements.

As a consequence of the relative character of articulation scores, all comparisons of communication devices should be made with conditions kept as uniform as possible. A comparison of two microphones, for example, should be made only with regard to the specific conditions governing the particular comparison. If both microphones are retested with a different amplifier, or with different earphones, or in a different noise, they may well give different results. Similarly, in comparing listeners, it is important to make judgments about their relative abilities to hear words or commands only on the basis of data obtained under comparable conditions.

Some of the factors which influence articulation test results are important only under special circumstances, while others should be considered in every articulation experiment. The following list attempts to summarize the more obvious factors which may affect articulation results.

Announcer	1. Quality and intensity of voice.
	2. Correctness of pronunciation.
	3. Manner of holding microphone, etc.
Speech	4. Phonetic composition and item difficulty.
Microphone	5. Frequency-response characteristics.
	6. Nonlinear distortion.
	7. Efficiency and impedance.
	8. Behavior at different altitudes.
	9. Directionality (shielding of microphone from noise — signal-to-noise ratio).
Amplifier	10. Frequency-response characteristics.
	11. Nonlinear distortion.
	12. Input and output impedances.
	13. Gain.
	14. Peak power limitation.
	15. Shielding (noise pickup and feedback).
Radio Link	16. Overall fidelity (response characteristic).
	17. Signal-to-noise ratio.
	18. Loudness of side-tone channel heard by announcer.
	19. Overmodulation.
	20. Presence or absence of limiters.
Earphones	21. Frequency-response characteristics.
	22. Nonlinear distortion.
	23. Efficiency.
	24. Behavior at different altitudes.
Earphone Mounting	25. Design of helmet and headband.
	26. Acoustic seal at the ear (insulation against noise).
	27. Air volume under receiver.
Listener	28. State of hearing (deafness).
	29. Masking of speech by noise entering ear.
	30. Basic ability to understand speech when distorted and masked.

This list of factors is by no means exhaustive. It is indicative, however, of the extensive matrix of parameters affecting any attempt to evaluate in a quantitative way the relative merits of communication systems, the effectiveness of noises in masking speech, the audibility of words and commands, or the relative abilities of communications operators.

5.1 THE FORMAL ARTICULATION TEST

Because of the complex multitude of variables involved, proper experimental control of all the relevant conditions presents a formidable problem. Although short cuts have been devised for greater economy of time and personnel, the most valid and reliable results can be obtained only by the formal articulation test. The formal articulation test involves the use of carefully selected test materials, the elaborate control of individual differences in talkers and listeners, a relatively permanent testing installation, and systematic variation of the intensities of the speech and noise employed. Abbreviated test procedures necessarily slight one or more of these aspects.

The following discussion summarizes the extensive experience of the Psycho-Acoustic Laboratory in using this formal method of testing.[5]

5.1.1 Test Material

Since the nature of the spoken items helps to determine the resulting articulation scores, the test material must be carefully selected. For most testing purposes the speech sounds used should be representative of conversational speech. Furthermore, for economy of effort and time, it is important to group the speech units used for the articulation test into convenient test lists, each list as difficult as each other list. When lists of comparable difficulty are used, differences in articulation scores obtained with two different microphones may be interpreted as due to the differences in the instrument rather than to differences in the difficulty of the list. Since the difficulty of a test list is determined not only by the fundamental speech sounds but also by numerous psychological factors, it is necessary to demonstrate by actual test that each test list is as difficult as each other list.

The desirability of a proportional representation in the test lists of the sounds which occur in everyday speech stems from a consideration of the problem of *validity*. A microphone which passes only one type of sound might test well with a list of words containing only such sounds, but the test would not be a valid indication of the usefulness of this instrument for ordinary conversation. It is possible to state approximately the relative frequencies of occurrence of sounds in "average" speech, and to approach these frequencies in the distribution of sounds among the lists of an articulation test. A standard reference on the distribution of speech sounds may be found in the literature.[1] In general, as the number of different speech sounds in the test lists is increased and as the relative frequency with which each sound occurs approaches that found in conversational speech, the more valid will be the articulation results.

For practical applications, it is usually desirable to obtain the articulation, not of the individual speech sound, but of combinations of the sounds. When combinations of speech sounds are employed, however, the articulation score depends not only upon the difficulty of the individual speech sounds but also upon the particular combinations used. In general, speech sounds may be combined in three different types of test items:

1. Single syllables made up of meaningless combinations of speech sounds.
2. Meaningful words given out of context as isolated units.
3. Meaningful phrases or sentences in which there are contextual relations among the words.

The principal differences among these three classes depend upon the psychological factors of meaning, inflection, rhythm, etc.

When it is desirable to determine accurately the particular speech sounds transmitted by an interphone, nonsense syllables are superior to words or sentences as test items, because with this type of material, the clues otherwise provided by the meaning of the combination are minimized. The use of nonsense syllables, however, requires that the testing crew be thoroughly trained. The announcer must correctly pronounce the speech sounds and the listeners must record with phonetic symbols the sounds they hear. When words or sentences are used, on the other hand, the training program is far less arduous.

The extent to which word articulation scores

are influenced by meaning, inflection, etc., depends largely upon the particular words chosen for the test. Short words which provide less opportunity for the operation of such factors are usually missed more frequently than long words. When sentence lists are scored in terms of the meaning conveyed these psychological factors are still more important. For this reason, sentence articulation is typically higher than word or syllable articulation. Figure 1

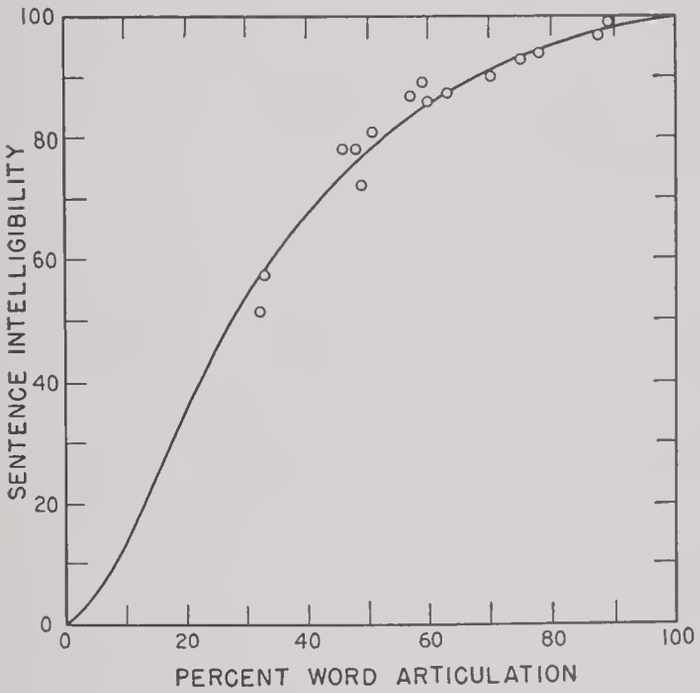


FIGURE 1. Lists of words and lists of sentences were compared for intelligibility under a wide variety of conditions. Each point on the graph represents 1 test of 100 words and 1 test of 50 sentences. The curve passing through the data was derived from results obtained at the Bell Telephone Laboratories.

illustrates the type of relation obtained experimentally between word and sentence articulation. This relation shows that sentence articulation is higher than the corresponding word articulation.

In addition to representing conversational speech by lists of equal difficulty, the test items must be selected in such a way that the distribution of item difficulty in each list will make possible a sensitive measuring instrument. The following hypothetical example illustrates the importance of the distribution of items with respect to their inherent difficulty to the listener. In this example, schematized in Figure 2, three different test lists are used in order to compare two interphone systems. Each of these test lists is comprised of twenty items of *known* difficulty. The items in the first list are uniformly distributed along the hypothetical scale of difficulty, as is shown in the figure. Ten of the items in the second list are difficult and ten are easy. The third test list includes twenty items of intermediate difficulty. Figure 2 also shows the articulation scores that would be obtained with two interphone systems, A and B, tested with each of the three test lists.

Although the first test list would provide a means of distinguishing between the two communication systems, this list includes *relatively* few items which are heard correctly over one of the communication systems but missed over the other system. Those items which are always correctly recorded or always missed when read over the systems tested do not contribute to the sensitivity of the articulation test. Such items are "deadwood," and if they are eliminated from the test list, time and effort will be saved.

The second list, with only very hard and very easy items, is of little or no use in distinguishing between the two systems, A and B. It is not an adequate test because it does not measure the difference in intelligibility provided by the two interphone systems. It says, in effect, that the two systems are identical. Such a test list would be of value only when very good or very poor communication systems are tested.

For the particular task of appraising these two interphones, the third test list, comprised only of items of intermediate difficulty, is the best list. This list is economical in that it contains few items of dead weight. It provides a sensitive measure of the relative merits of systems A and B.

The foregoing example illustrates three important requirements for the distribution of test items with respect to difficulty. If test lists are to be sensitive to small differences in intelligibility and convenient for use, the test items must be fairly closely distributed along a scale of difficulty. Secondly, those items which under the conditions of the tests are always recorded correctly or always missed are dead

weight and may well be omitted from the test lists. Finally, the number of test items in a list should be sufficiently large and the distribution of difficulty sufficiently wide to embrace the requisite range as determined by the types of communication equipment under study.

the number of tests. For this reason it is important to have available a large number of equivalent lists of test materials and to choose judiciously the number to be administered at any one time.

As a practical consideration, it should be

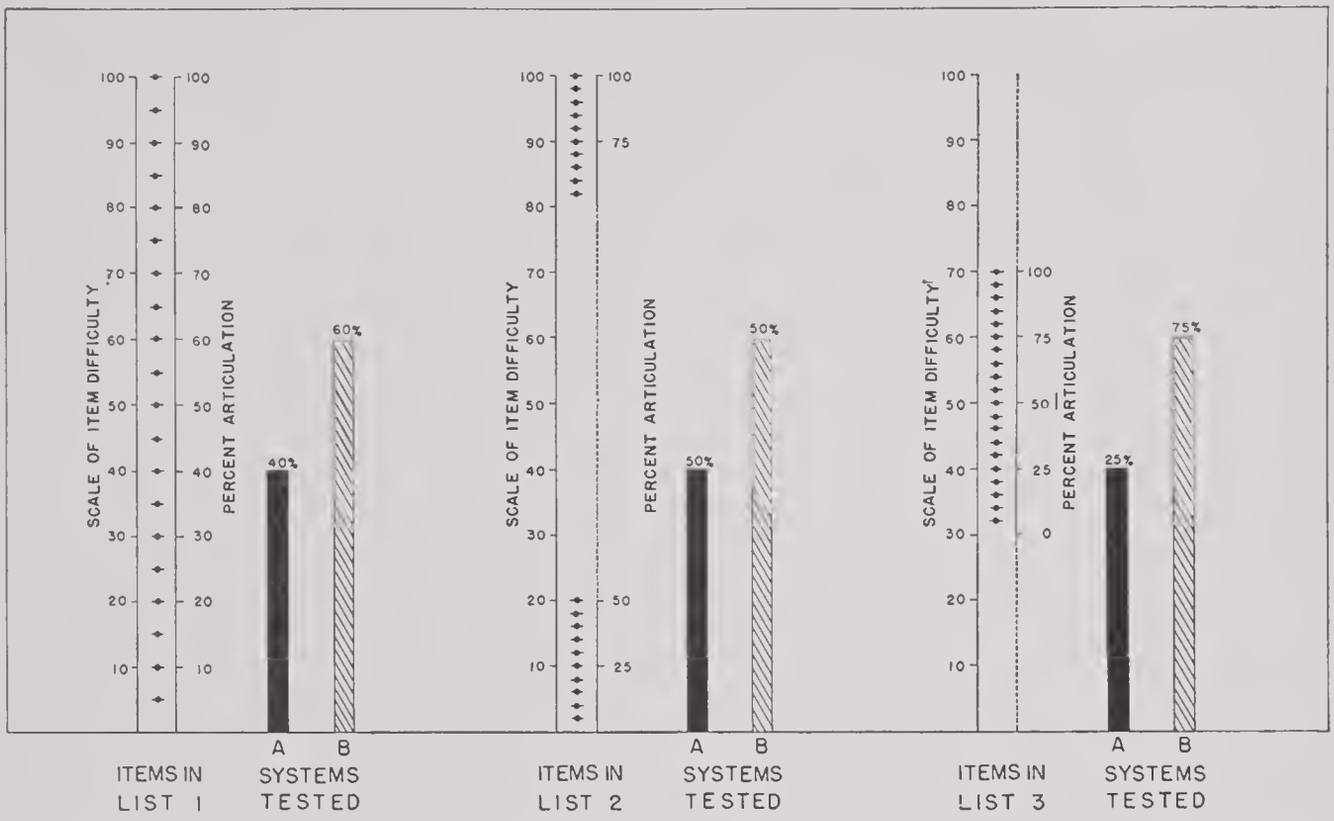


FIGURE 2. Illustrating how two communication systems would be appraised by 3 idealized lists of 20 words each. The distribution of the items with respect to their inherent difficulty is indicated for each list. Each system is represented by a bar whose height, against the scale of difficulty, shows what items the system is able to transmit intelligibly. The resulting articulation score is shown above each bar.

It should be pointed out that in terms of reliability alone, i.e., the ease with which an articulation score could be repeated on a second testing, the second list shown in Figure 2 is the most reliable and the third is the least reliable. With the second list a score of 50 per cent would be obtained regardless of small random fluctuations in the subtle factors which always affect articulation scores. Scores obtained with the third list would be quite sensitive to these random factors. It is plain, therefore, that a reliable, repeatable test is not necessarily a good one, and it is also obvious that a sensitive test is apt to be inherently unreliable.

Fortunately, the reliability of an articulation score can always be improved by increasing

noted that in any actual list of test items the distribution of difficulty would probably never follow the ideal scheme presented in Figure 2. Instead, the items would tend to follow a bell-shaped distribution along a scale of difficulty. For this reason, articulation tests are ordinarily not uniformly sensitive over the complete range of possible scores. The greatest sensitivity usually results when the testing conditions are so adjusted that scores near 50 per cent are obtained.

The actual task of assembling lists of materials for articulation testing proved to be both interesting and endless. Following the first extensive word lists compiled by the Psycho-Acoustic Laboratory in 1941, there was continual tinkering, testing, and trying by one or

another research group in an effort to improve old lists or devise new ones for special applications. The story of the design of nonsense syllable lists, monosyllabic word lists, spondaic word lists, and sentence lists has been discussed in detail elsewhere,[5] along with lists of test materials of these four types which were designed with relevant considerations in mind.

In administering these materials, some consideration must be given to the use of a carrier sentence and to the time interval between test items. In word or syllable articulation tests each item is usually read as part of a sentence. For example, in the sentence, "You will write *car*," the word *car* is the only word which the listener is required to record.

A carrier sentence is desirable for several reasons. (1) The listener is prepared for the presentation of the test item, and variability in the articulation scores due to inattention or distraction is reduced. (2) If carbon-button microphones are used, the carrier phrase preceding the test items serves to agitate the particles of carbon and reduce the variability inherent in such microphones. (3) The carrier sentence permits the announcer to modulate his voice so as to keep the level of his voice even from word to word.

In some experiments a carrier sentence is selected to fulfill a more specific purpose. When articulation tests are conducted at high altitudes, for example, it may be desirable to use a fairly long sentence. In this way, the effect on articulation due to the difficulty of breathing at high altitudes may be assessed. One such carrier sentence that has been used successfully is "One, two, three, four, five, *test item*, seven." All the words in this sentence, including the test item, should be spoken evenly and with one breath. No attempt should be made to accentuate the test item.

For most purposes, the carrier sentence and not the test item should be used to monitor voice level. No attempt should be made to compensate for the typical differences in the speech power used in pronouncing the different sounds in the test item. When only the carrier sentence is monitored, the test item should be spoken with the same general effort as the rest of the carrier sentence.

The time interval between test items should be just long enough to enable the listeners to record the test item. With a well-trained crew the average rate at which words or syllables may be read is one item every three to four seconds. With this interval it is possible to conduct from 25 to 35 tests of 50 words each in a three-hour testing session. With a session this long, two to four 15-minute rest periods should be given.

5.1.2 Personnel

The intelligibility of speech heard over a communication system depends upon the vocal quality of the announcer and the listening abilities of the listener, as well as upon the quality of the physical instruments used. Since large individual differences in talkers and listeners are to be expected, a number of factors should be considered in selecting the testing personnel. Announcers should be selected who are able to enunciate the fundamental speech sounds in a "normal" manner, where the criterion for normality is one of common sense. Listeners having the equivalent of a high school education and whose ages range from 17 to 30 years are usually suitable for this work. Sex differences appear to be unimportant so far as listening ability is concerned. For most purposes, the listener should have reasonably normal hearing, as determined by standard audiometric procedures.

Articulation scores obtained with inexperienced listeners show improvement with practice. Figure 3 shows a typical learning curve obtained under severe acoustic stress. After two to three hours of practice each day for a period of five successive days, this crew had reached a relatively stable level of performance under the particular conditions chosen for the tests.

From the thousands of articulation tests that have been conducted in the Psycho-Acoustic Laboratory with various numbers of talkers and listening crews, under a great variety of testing conditions, it is possible to make an approximate estimate of the optimum number of listeners, talkers, tests, and test words for

the type of test situation most frequently encountered. When the time available for testing is limited, it is usually most efficiently used by tests of about 50 words which, for purposes of administration, may be divided into two lists

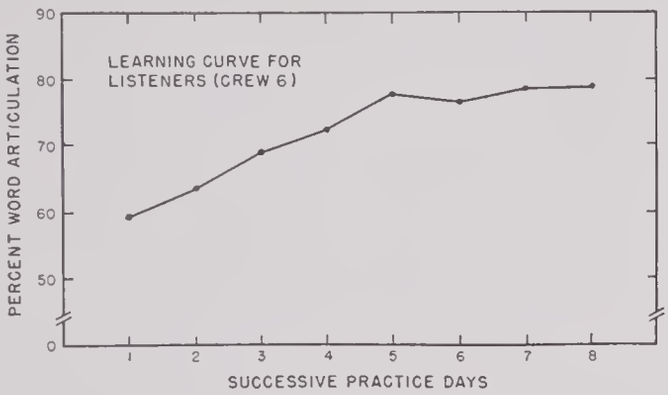


FIGURE 3. Each point on the curve represents the average score on 12 tests for a crew of 10 listeners. The tests were read over an interphone system by three well practiced announcers. Both the announcer and the listeners were in an ambient airplane noise (115 db).

of 25 words each. Since this length has been found convenient both for administration and scoring, the estimates of the optimum number of listeners, talkers, and tests are made on the assumption of 50-word lists.

The score of a single listener, due to his own variability, may vary 7 or 8 per cent from one test to another under the same conditions. This variation can be effectively reduced by reading tests to a group of listeners rather than to a single listener and by using the mean articulation score of the group. The mean articulation score obtained from a well-trained crew of six to eight listeners will be sufficiently stable for most experiments.

Fatigue from long hours of listening to word lists, particularly in the presence of intense noise, is a factor which might be expected to introduce systematic errors. However, experiments designed especially to measure the effects of fatigue on articulation scores demonstrate that under typical conditions of testing and in the presence of an intense ambient noise, little or no change in the articulation scores results from several hours of testing.

The variability of the talker is a much more important source of instability in the final score. The effect of this variability can be reduced only by having each talker read a number of tests and by averaging the mean articulation scores for the several tests. The average of the mean articulation scores of a set of six 50-word tests read by a single talker to a crew of eight experienced listeners under conditions permitting about half the words to be heard correctly will probably be as stable a value as will be necessary for most purposes. As the number of tests is increased beyond six, under these conditions, the stability of the average increases only slightly with each additional test.

Increasing the number of talkers and listeners has two effects: it improves reliability of the test and it makes the results of the test more representative of the results that would be obtained from an unlimited population. Against these advantages must be weighed the cost in time and effort. The point of diminishing returns is reached and passed if the numbers are compounded too rapidly.

In a great many testing situations, the number of different talkers used is not important. With some kinds of equipment, however, it is important to use a wide variety of talkers in order to obtain representative results. A man with a thick, fat neck may be a superior talker with an ordinary hand-held microphone, but he may be almost unintelligible over a throat microphone because of the padding on his throat. In general, it is with tests of microphones that the problem of selecting representative talkers is most pressing.

Variability due to inexperienced announcers and listeners can often be reduced by a short period of instruction. The way in which the testing personnel use the microphones and headsets under test are important conditions of an articulation test. In order to control the speech input to the microphone throughout any one test it is necessary to keep the face of the microphone at a constant distance from the lips of the announcer. Figure 4 shows the results of a simple experiment illustrating the importance of instructing the announcer regarding this factor.

The instructions given to the listener regarding the use of headsets also depend upon the purpose of the experiment. When the listeners take the tests in an intense ambient noise, the

acoustic seal afforded by the headset is an important factor.

Finally, the listeners should be instructed to record a response for every item called by the

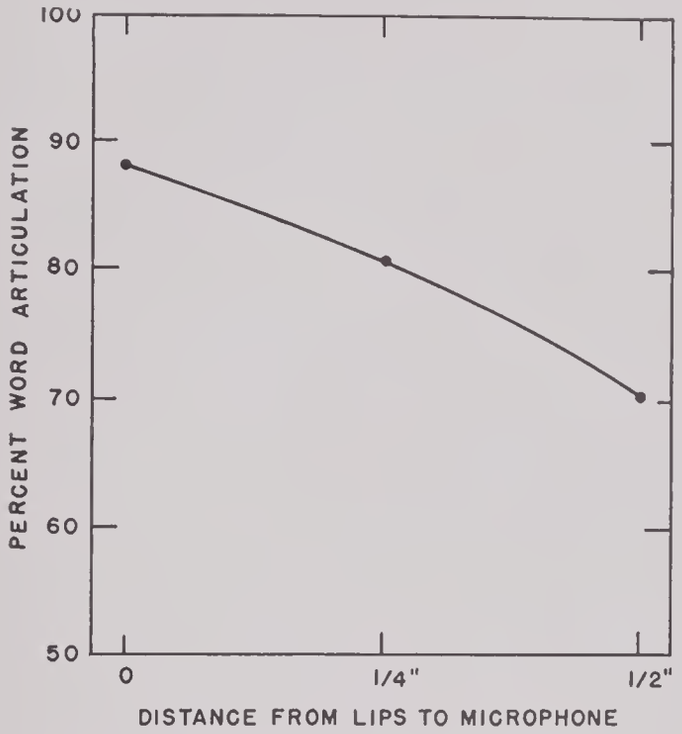


FIGURE 4. Showing the effect on communication of varying the distance of the microphone from the lips. The tests were conducted with the announcer and crew of listeners in an airplane noise (115 db). Each point is based upon the average of tests read by three announcers.

announcer. It has been shown in many experiments that the listener records correctly some test items even though he maintains that he is only guessing.

5.1.3 Equipment

In comparing two devices with respect to the intelligibility of speech transmitted over them, it is important to consider the other parts of the communication system used for the tests. For example, suppose that of two microphones tested, one microphone does not transduce speech frequencies above 2,500 c and the other transduces all the important frequencies of speech. If earphones which do not transduce speech frequencies above 2,500 c are used for these tests, little or no difference would be found in the articulation scores provided by the two types of microphones. If, however, earphones having a wide frequency response were selected for this comparison, a higher articulation score would be obtained with the microphone which transduces all the important frequencies of speech. It may be stated as a general rule that the final evaluation of an instrument cannot be made apart from a consideration of its associated equipment.

One of the most convenient equipment arrangements provides a high-fidelity system with sufficient flexibility to permit the substitution of the equipments to be tested for the corresponding components in the testing installation. Sometimes a complete system from microphone to headset is to be studied, and in such a case the testing system is not needed. More often, however, only one or two items of a communication system are tested, and in this case the use of complementary equipment of good quality is the easiest to specify and shows the clearest picture of the component's performance.

5.1.4 Noise Conditions

In most communication systems the ambient noise which reaches the ear is either picked up by the microphone or enters by way of acoustic leaks in the headphones or helmet which the listener wears in a noisy environment. The kind and the amount of ambient noise which enters the system through the microphone depends upon three main factors: the intensity and spectrum of the ambient noise (see Chapter 2), the amount of acoustic shielding which the construction and mounting of the microphone provide (see Chapter 9), and the response characteristic of the microphone itself. The acoustic seal provided at the ear by various headsets and helmets also varies over a wide range. Hence, the intensity and spectrum of the interfering noise leaking into the ear by this path may vary greatly.

In addition to ambient noise, *electrical* noise interference is often picked up by radio equipment. This electrical interference may arise from atmospheric static, from adjacent installations (notably radar), or from deliberate jamming attempts by the enemy.

The noise which reaches the listener's ears tends to mask portions of the speech signal. The amount of masking produced depends primarily upon the intensity and the spectrum of the interfering noise. By way of illustrating the

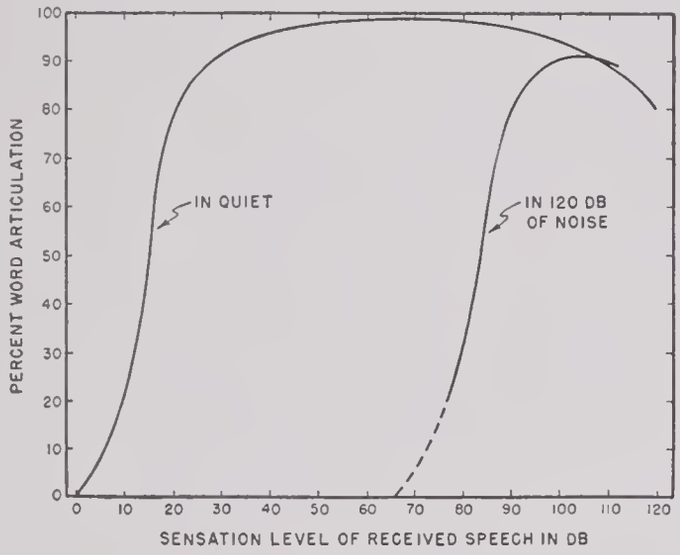


FIGURE 5. Effect of loud ambient noise on the relation between per cent word articulation and received speech intensity. When the announcer and the listeners are in an intense ambient noise (120 db), the range of levels of received speech available for intelligible communication has been reduced by about 65 db. (See reference 3, p. 272.)

effects of a particular noise, Figure 5 shows the extent to which an intense ambient airplane noise (120 db) reduces the range of levels of received speech available for intelligible communication. In order to obtain the same articulation score in 120 db of this noise as would be obtained in the quiet, it is necessary to increase the intensity of the speech signal approximately 65 db.

The particular noise conditions used for testing will depend upon the purposes of the experiment and upon the anticipated uses of the equipment under test. In testing equipment for use in high ambient noise fields, there are four permutations of noise and quiet under which articulation tests can profitably be conducted:

1. Quiet-to-quiet: both announcer and listeners in quiet (Q-Q).

2. Quiet-to-noise: announcer in quiet, listeners in noise (Q-N).

3. Noise-to-quiet: announcer in noise, listeners in quiet (N-Q).

4. Noise-to-noise: both announcer and listeners in noise (N-N).

Each of these arrangements has its analogue in one or another mode of field operation.

The situation is further complicated when an electrical noise is added to the interference produced by an ambient noise. Tests conducted under these conditions indicate that the articulation scores obtained are dependent upon the predominant source of noise. Thus, for example, if a radio receiver is being tested for its susceptibility to atmospheric static and if an ambient noise is also used in the test, the ambient noise controls the articulation scores over the lower portion of the range of speech intensities. When the speech signal and static are well above the level of the ambient noise, however, the amount of interference is dependent upon the strength of the electrical noise. It is generally advisable, therefore, to simplify the experimental situation when the effects of electrical noise are under consideration by eliminating the use of ambient noise for acoustic stress.

5.1.3 Speech Intensity

Signal level is one of the most important determinants of the intelligibility of speech. For convenience, signal level in interphone communication may be considered a function of five independent variables:

1. The voice level of the announcer.

2. The efficiency with which the voice of the announcer is coupled to the microphone.

3. The frequency response and sensitivity of the microphone.

4. The gain of the associated amplifiers.

5. The frequency response and sensitivity of the earphone when it is coupled to the ear.

In experiments designed to study the relation between articulation and the level of received speech, it is more practicable to vary voice level or the gain of the amplifier than any of the other three variables. As a matter of actual practice, it is amplifier gain that is varied in most experiments. Articulation measurements made under these conditions are commonly referred to as gain functions. In

determining gain functions, the announcer should maintain his speaking level constant throughout any one experiment. Both visual and auditory means of indicating voice level should ordinarily be provided. The output voltage of the microphone being used may be employed to monitor the voice when the noise level due to the microphone pickup is about 5 db or more below the speech level. When the speech-to-noise ratio measured in terms of the voltage developed by the microphone is less than 5 db, this method is inadequate. Furthermore, carbon microphones do not provide a correct indication of changes in voice levels under typical conditions of testing. At high voice levels, these microphones distort the speech signal and an effective compression occurs. Consequently, changes in the output voltage of some, if not all, carbon microphones are not proportional to changes in voice level (see Figure 6).

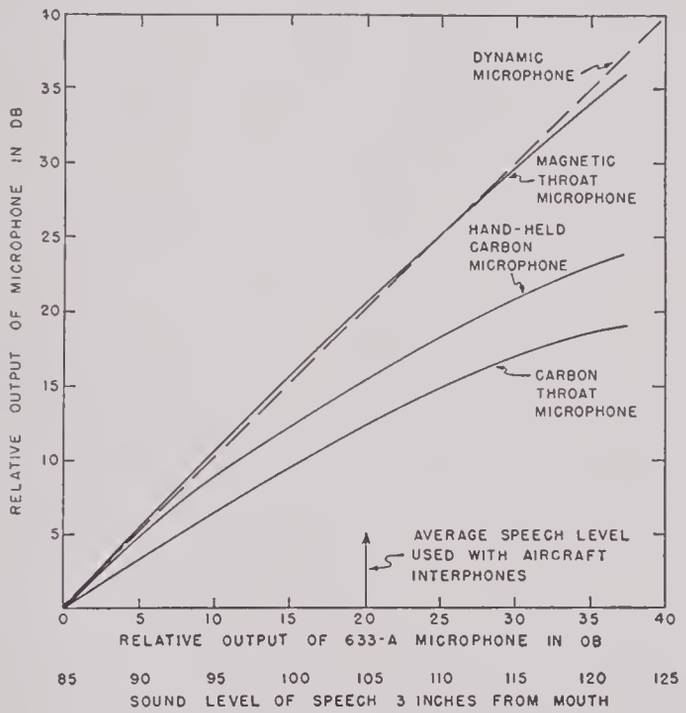


FIGURE 6. Showing the relation between voltage output of various microphones as a function of speech input intensity. Curves represent average results for three speakers, each of whom read 10 words at five voice levels when using the following types of microphones: (1) Two T-30-P and two T-30-V carbon throat; (2) three sonotone magnetic throat; (3) two T-17 (one Kellogg and one Universal) carbon hand-held; (4) one 633-A dynamic. The curves have been made to coincide at the lowest speech level tested. Indicated by the arrow is the average speech level used by a crew of enlisted men in flight tests at Eglin Field, Florida. All output voltages were measured with a VU meter.

In most experiments a monitoring system which is independent of the system under test is desirable. By using a *magnetic* throat microphone with an amplifier and output meter, a monitoring system is obtained which is satisfactory over a wide range of experimental conditions. As shown in Figure 6, changes in voice level are quite accurately indicated by the voltage output of a magnetic throat microphone. The chief deficiency of this method is the difficulty in placing a throat microphone in the same position on the throat from test to test, but this objection can be overcome by the exercise of sufficient care.

When articulation tests are conducted at only one speech level it is frequently useful to know the sensation level of the received speech signal. The sensation level of a given sound is the number of decibels that the sound is above its normal threshold of audibility. The sensation level of speech may be determined by attenuating the speech level until a listener can detect the presence of about 50 per cent of the speech sounds without identifying any of them. The amount of attenuation in decibels required to obtain this threshold is used to determine the sensation level of a given sample of speech.

5.1.6 Statistical Methods

If a test is repeated a large number of times under the same conditions, scores of various magnitude will be obtained. A frequency distribution of these scores typically shows that the scores occur most frequently around some central value, and scores which deviate markedly from this central value are relatively infrequent. The dispersion of the distribution of average scores obtained under the "same" conditions will depend to some extent upon how well the experimental conditions have been controlled and also upon the number of listeners and the length of each test list. Under typical conditions of testing in the Psycho-Acoustic Laboratory, it has been found that the standard deviation (root-mean-square) of a distribution of articulation scores obtained

by reading different 50-word lists under the "same" conditions is about five score units.

Because of this variability among individual tests, it is highly desirable to base conclusions regarding articulation efficiency on the average of several repeated tests. In general, the mean of a group of tests is far more stable than is the score for a single test. This stability increases as the square root of the number of tests, so that by reading four word lists, for example, the expected variability of the mean score can be reduced by a factor of two.

The basic principle in experimental design consists in controlling at known values as many variables as possible and in so arranging the remaining factors that their effects are made as random as possible. By this procedure it is possible to interpret the central or average articulation scores in terms of the known conditions, and to determine the limits of accuracy imposed by the errors of random sampling. The dispersion of the articulation scores then becomes a measure of the precision of the experiment, and an estimate of the error of measurement may be based upon this dispersion. It is then a straightforward matter to assess, by conventional statistical procedures, the reliability of the differences obtained between the scores for different instruments, personnel, testing conditions, etc. Such procedures can be found in any standard text on statistics.

In general, the most direct way to increase the precision of an experiment, in order that small differences in articulation may be reliably assessed, is to increase the number of independent articulation tests. When the amount of time available for testing is limited, however, more reliable differences will be obtained with short test lists than with longer test lists. Those errors which tend to raise or lower the average articulation score throughout the reading of a given test list will have less influence when a large number of short tests is used. Orders and arrangements can then be counterbalanced, since with short tests it is possible to sample a larger number of announcers in a given time, and thereby make the test more representative of conditions that might be encountered in practice.

Finally, if a statistically significant difference is not obtained in a given experiment, it cannot be concluded that there is no difference between the devices tested. An experiment in which more tests are conducted, or an experiment in which the conditions are more precisely controlled, may show a reliable difference among the communication devices compared.

5.2 ALTERNATIVE METHODS OF EVALUATION

The formal articulation test based on groups of announcers and listeners may prove too cumbersome and inefficient when an articulation study involves numerous permutations of experimental conditions. It is then expeditious to devise short-cut methods. Many schemes are, of course, possible and several variants have been tried from time to time.

5.2.1 Abbreviated Testing Methods

An interesting example of an efficient procedure is the check-list method. This method uses one or more listeners and the talkers are replaced by their recorded voices. The list of words are recorded in several different scramblings, and the listener follows their presentation with the aid of a check list, uncovering each word after he hears it spoken. He then indicates by a check mark whether or not he has heard the word correctly. Since the listener must establish his own criterion for "hearing" a word, training is necessary before consistent results are obtained.

The obvious disadvantage of this method is that the data reflect the performance of only one listener and are to some extent dependent upon his idiosyncrasies of judgment. The method has been used, however, with as many as four listeners simultaneously. Increasing the number of listeners does not affect the amount of time which can be saved by this method. Since the listener, instead of writing down the word that he hears, merely checks his correct responses, he is able to work at a fast pace and the speed-up in the articulation testing is considerable.

If only the total scores, and not the results for individual items are desired, it is possible for the listener to follow the check list using a simple manual counter which he punches after he has decided that he heard the word correctly. If lists of 100 words are used, the counter reading can then be taken as the percentage articulation score.

The same considerations regarding the selection of talkers and of test material apply for the check-list method as for the more formal articulation testing methods. For the purposes of this laboratory, 100-word lists have been compiled in which 15 vowels and diphthongs are represented by six words each. The remaining ten words of each hundred are used to sample some of the consonants.[5a] Especial care must be used to secure high-fidelity phonographic recordings of the talkers' voices.

5.2.2 Comparisons Based Upon Subjective Appraisal

In an articulation test the valuation of the intelligibility of speech is based upon the number of test items correctly recorded by the listener. In this kind of test, the listener is not required to appraise the quality of the speech, but merely to record the speech sounds that he hears. By contrast, methods of subjective appraisal require the listener to evaluate the quality of the speech itself. One variation on the subjective procedure is the method of rank order. This requires the listener to judge which of two or more samples of speech is the more intelligible. Another variation is the rating-scale method, which requires him to describe a sample of speech in terms of its quality. The listener must make this latter judgment in terms of a standard sample of speech provided for him, or in terms of his previous experience in evaluating speech.

If only an approximate assessment of the intelligibility of a speech sample is desired, these subjective methods may be found useful. They can be employed to especial advantage in the preliminary evaluation of a communication system. Also, the experimental evidence indicates that the methods may be reliably used to rate the intelligibility of different talkers (see Chapter 14).[4] However, for the accurate measurement of small differences in speech quality, more formal articulation testing methods should be employed.

5.2.3 Threshold Methods for Evaluating the Intelligibility of Speech

It is frequently desirable to know the conditions under which speech can just be heard at some arbitrary level of clarity. Three of these threshold methods will be described:

1. The threshold of detectability.
2. The threshold of perceptibility.
3. The threshold of intelligibility.

To determine the *threshold of detectability* (sometimes called the threshold of audibility)

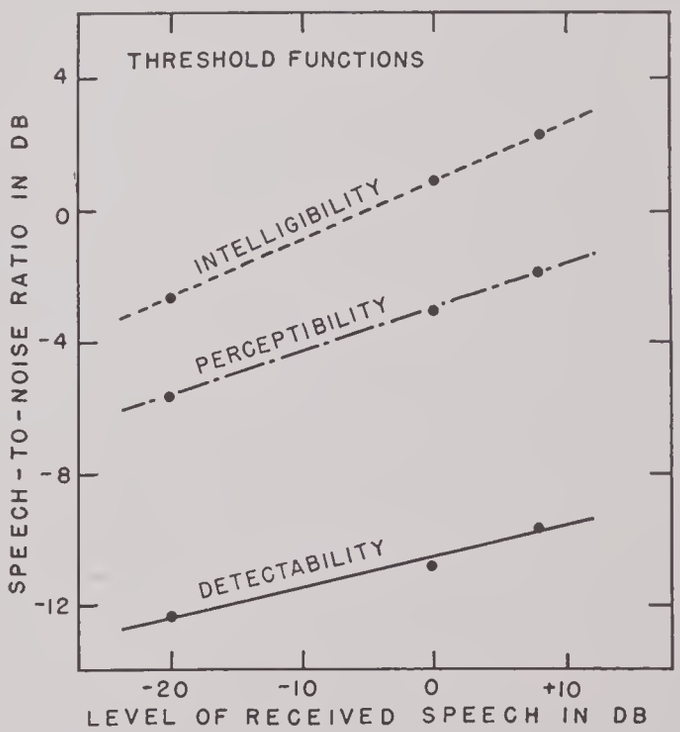


FIGURE 7. Showing the speech-to-noise ratios required to obtain three types of thresholds. White noise (random spectrum) at various levels was mixed electrically with the speech. The abscissa level of 0 db corresponds to a sound-pressure level of about 108 db at the listeners' ears.

the listener adjusts some variable (usually the speech level or the level of a masking noise) until he is just able to detect the presence of speech sounds about half the time. At this

threshold level he will ordinarily be unable to identify any of the sounds themselves.

The *threshold of perceptibility* can be used to determine that condition under which the sounds heard at the threshold of detectability begin to be perceived as words. In determining this threshold, the listener adjusts some variable until, in his judgment, he is just able to understand with considerable effort the gist of the connected discourse read to him. If the speech were made intelligible he could, with little or no effort, understand it, and if the speech were made less intelligible, he could not perceive a sufficient number of words to allow him to follow the main ideas of the passage read to him.

In determining the *threshold of intelligibility* the listener adjusts some variable, until, in his judgment, he is just able to obtain without perceptible effort the meaning of almost every sentence and phrase of the connected discourse read to him.

The results of an experiment designed to evaluate these three threshold methods are shown in Figure 7. The level of the received speech was held constant and the listener adjusted the level of an interfering random noise until he obtained each of the thresholds. This procedure was repeated using two other levels of received speech. The results show that, for this type of noise, the speech-to-noise ratio required for the threshold of perceptibility is about 7 db higher than that required for the threshold of detectability. A further increase in the speech-to-noise ratio of only 4 db is required to reach the threshold of intelligibility where the listener is able to understand easily every sentence.

Chapter 6

INTELLIGIBILITY OF SPEECH: SPECIAL VOCABULARIES

EXPERIENCE IN ARTICULATION testing quickly verifies the common-sense notion that some speech sounds, and consequently some words, are more communicable than others. The intensity and spectra of the component speech sounds, the number of syllables and the relation of a word to other words combine to produce wide differences in recognizability. Because of these differences, haphazard selection of words for use in the din of mechanized war will not result in the most effective communication. Consequently, the determination of the factors contributing to the intrinsic recognizability of a speech signal and the provision of vocabularies of words easily recognizable in noise became important research endeavors.

6.1 METHODS OF TESTING

It is possible to establish the differences in the recognizability of individual words by means of an adaptation of articulation testing methods. The procedure differs from the standard articulation tests primarily in the analysis of the results; words are compared rather than communication systems.

The series of experiments directed at this problem used from four to seventeen speakers and from eight to sixteen listeners. An attempt was made to select speakers representative of a wide variety of dialects. Different communication systems were exposed to ambient and electrical noises and the ratings of the words obtained in one situation were carefully compared and correlated with the ratings obtained with different talkers, different systems, or different noises.

Caution must be employed, however, in interpreting these results. For the practical purpose of selecting a highly intelligible vocabulary, the important point is not the absolute score of each word, but the comparative score of the word. Since in each experiment the ratio of signal to masking noise was set to give optimum discrimination among the words being tested, the absolute score for each word is relative to the signal-to-noise ratio of each separate experiment. In addition, even the relative standings of the words tested in an experiment of this type must be interpreted with regard to the speakers and listeners participating in the particular experimental procedures, and finally, the ratings must be regarded as to some extent dependent upon communication systems employed. If all of these factors are taken into account, however, it is reasonable to expect that words selected on the basis of objective tests rather than at random or according to the whim of radio-telephone operators can aid substantially in overcoming the obstacles to successful communication in noise.

6.2 ALPHABETIC EQUIVALENTS

In oral communication of military messages, the "phonetic alphabet" or words used to identify the individual letters of the alphabet plays a very important part. This alphabet is used over the radio-telephone not only to spell out difficult words, but also to transmit cryptograph messages and standard commands. It is essential to the success of many operations, therefore, that all 26 of these equivalents be instantly and unambiguously identified even against the intense noise of mechanized battle.

Consideration of the conditions under which alphabetic equivalents are used in actual service made it apparent that a satisfactory list must meet the following criteria:

1. All the words in the list should be recognizable when heard over an intercommunication system against a background of severe noise. This, of course, is the most essential criterion. The words should be identifiable (1) by listeners who are comparatively unfamiliar with them, (2) when the words are read over a representative variety of communication systems, (3) when the listeners represent a wide range of educational levels, and (4) when the speakers and listeners represent different re-

gional dialects in both the United States and the British Empire.

2. All the words should be in common use, easy to learn, and of the utmost brevity compatible with a high degree of recognizability.

3. None of the words should be geographical or place names, since these might be misconstrued as forming part of an operational command.

4. None of the words ought to possess a facetious connotation or be of negative morale value. For example, the British use of "nuts" for "N" would seem unsuitable for American adoption.

In all, approximately 300 different words were tested, drawn from the following sources:

1. The LCC (British) list.
2. The USA (American) list.
3. The CCB list, composed of words suggested from time to time by members of the Combined Communications Board.
4. The various words used by the American Telephone and Telegraph Company, the Bell Telephone Company of New Jersey, RCA Communications, Western Union, and the International Telecommunications Convention.
5. The NDRC list, composed of the most intelligible words suitable for use as alphabetic equivalents, selected from a list of 2,400 common English monosyllables and dissyllables which had been used in articulation testing of communication equipment in the Psycho-Acoustic Laboratory over a period of eight months.

In order to determine the extent to which these 300 words satisfied the criteria for alphabetic equivalents, seven tests were conducted. In all tests, both speakers and listeners sat in 120 db of ambient noise. The noise simulated the spectrum typical of an acoustically untreated, twin-engined bomber.

In the light of the results of these experiments, the following words are recommended as the best equivalents for each letter of the alphabet. Words in parentheses represent a second choice which is only slightly inferior to the first choice. The words in Column II compose the phonetic alphabet which has been adopted for combined United States-British radio-telephone procedures.

I Best Equivalent Tested	II Equivalent Adopted by U. S. and British Board
Ablaze	Able (Affirm)
Boycott (Bugle)	Baker
Crayon (Chowder)	Charlie
Dinette (Dog)	Dog
Eardrum	Easy
Flower (Fortune)	Fox
Golden	George
How (Hypo, Headline)	How
Interrogatory (Ideal)	Item (Interrogatory)
Joyful	Jig
Knicknack (Keyhole)	King
Limeade	Love
Mary (Mike, Manhood)	Mike
Nowhere (Nylon)	Nan (Negat)
Oliver (Oatmeal)	Oboe (Option)
Piano (Parole)	Peter (Prep)
Queen (Queer)	Queen
Roger (Robust)	Roger
Sugar (Skillet, Supreme)	Sugar
Thyself (Trombone)	Tare
Umpire (Upstairs)	Uncle
Victory	Victor
William (Welfare)	William
X ray (Exhale)	X ray
Yuletide (Yellow)	Yoke
Zanzibar (Zebra, Zodiac)	Zebra

On the basis of the articulation test results, the adopted list is not, in various instances, the best list available. It must be remembered, however, that a number of factors other than recognizability must be taken into account in composing any set of military signal procedures.

6.3 ALTERNATIVE PRONUNCIATIONS OF NUMERALS

Numerals, like the phonetic alphabet, are such frequent and crucial components of oral messages that their instant recognition is essential to the success of many military operations. As normally pronounced, however, certain numerals are often confused with other numerals and words, even in the quiet. Against the intense noises of war such confusions increase, and some numerals become almost inaudible. To overcome these defects, various modified pronunciations of numerals have been proposed. The laboratory's problem was, therefore, to evaluate experimentally the comparative identifiability in noise of the various al-

ternative pronunciations of numerals. The pronunciations tested were drawn from the current United States and British pronunciations, the pronunciations employed by the Bell Telephone Company, and several other pronunciations proposed by the Psycho-Acoustic Laboratory.

Three experiments were conducted in order to establish a list of pronunciations for the single digit numerals in which, for U.S. and British speakers and listeners using a variety of local dialects, each pronunciation is the most identifiable of the various alternatives, and no pronunciation is readily confused with any other numeral in the list.

Both on the grounds of superior identifiability and lack of susceptibility to confusion, the following recommendations were made:

No.	Recommended	Acceptable	Not Acceptable
0	Ze-ro	Zero	Oh
1	Wun	Uh-wun	Wun-er
2	Too	Tuh-hoo	...
3	Thuh-ree	...	Th-r-eee; Three
4	Four	Fo-wer	...
5	Fi-i-v	Five	Fi-yiv; Fife
6	Six	...	See-yix; Sixer
7	Seven	Sev-ven	...
8	Ate	...	Ate-er
9	Niner	...	Ni-yen; Nine

Since these experiments were conducted and reported to the Services, the following pronunciations of numerals have been adopted for combined United States - British radio-telephone procedures:

Zero	Fi-yiv
Wun	Six
Too	Sev-ven
Thuh-ree	Ate
Fo-wer	Niner

With the exception of "fi-yiv," all these choices proved satisfactory in the audibility tests. Since "ni-yen," which had a strong tendency to be confused with "fi-yiv," has been changed to "niner," it may well turn out that the audibility of "fi-yiv" will be greatly improved.

6.4 TELEPHONE DIRECTORY NAMES

Telephone directory names make up another important component of standard military communication procedures. These are words used to identify telephone units during the course of a military operation. In the summer of 1942, a representative of the Office of the Chief Signal Officer requested the Psycho-Acoustic Laboratory to establish experimentally the comparative intelligibility in noise of the 500 five-letter telephone directory names used by the U. S. Signal Corps.

The telephone names submitted for tests had not been devised on the basis of intelligibility in noise, and it turned out that the different letters of the alphabet had varying numbers of satisfactorily identifiable names. Under each letter of the alphabet, even those words which could be regarded as satisfactory exhibit great diversity in recognizability. Also, since no name on the list was more than five letters in length, a great many of the most audible words in the language did not occur at all.

It would be possible, of course, to arrange the call names in order of identifiability and to devise directions for their use which insured the choice of the most audible words. The Psycho-Acoustic Laboratory has prepared such a listing with suggested instructions for its use.[2a] Even so, the list is imperfect and the instructions concerning its use are unnecessarily complicated and cumbersome. Consequently, it was suggested that the telephone directory names be either supplemented or replaced by words from a uniform list of 1,000 highly intelligible words described below.

6.5 VOCABULARIES OF HIGHLY INTELLIGIBLE WORDS

Each of the experiments discussed so far was intended to supply a small stock of speech signals for a single, specialized requirement. In the course of these experiments, it became evident that it would save considerable time and effort if a large scale experiment were to be conducted to provide an extensive general vo-

cabulary of highly intelligible words which could be drawn upon by all branches of the Armed Services to satisfy their special needs. From this vocabulary any branch of the Armed Services could select sets of speech signals to be used for military codes; for telephone directory names; for names of ships, planes, tanks, and infantry units; for standard operational commands; and for any other needs which might arise in the course of military communications. The allocation of words drawn from this vocabulary might replace the common practice of using speech signals devised more or less haphazardly without knowledge of their intrinsic intelligibility.

Experience in the field and laboratory has shown that such a vocabulary must satisfy at least three criteria.

1. All words must be easily identified in noise.

2. The words should not conflict with speech signals already in use for other purposes.

3. There should be a large number of names of commonly recognized classes and subclasses of objects (animals, plants, minerals, colors, etc.) to identify the interrelations among military units.

In order to provide a final list of 1,000 words, several thousand words were reviewed on the basis of evidence collected in previous experiments and then reduced to the final testing list of 1,600 words. Half these 1,600 words were words that had turned out to be the best in a list of the 3,000 most common monosyllables and dissyllables in the language; 400 were trisyllables and quadrisyllables selected from *Webster's Collegiate Dictionary*, and 400 were names of commonly recognized classes and subclasses of objects (animals, minerals, plants, colors, musical instruments, presidents, colleges, stars, Indian tribes, textiles, characters and books of the *Bible*).

The results of the tests and the recommended list of 1,000 words have been reported in detail elsewhere.[2] The vocabulary recommended by the Psycho-Acoustic Laboratory was later adopted by the War Department and reprinted in a War Department technical bulletin.[4] A large proportion of the recommended words are trisyllables because intelligibility tends to increase as the number of syllables in the word is increased. Unless there is a pressing need for economy of utterance, it is necessary to use a large proportion of trisyllables in order to make efficient communication possible in the most severe conditions of noise.

Upon conclusion of this research a request was received from the Signal Corps to supply 10,000 highly audible tactical call signs for use as call names for telephone units. Unfortunately the number of English words which are nongeographical in denotation, familiar to the average fighting man and also identifiable in noise, was exhausted by a list of 3,120 call signs.[1] This figure includes the 1,000 highly intelligible words discussed. Despite the unparalleled richness of the English language, it falls far short of providing 10,000 single words satisfying the criteria for military call signs.

To meet this deficiency, therefore, it is possible to devise a list of two-word combinations all of which will be both highly intelligible in noise and within the working vocabulary of the enlisted man. It is necessary to select the following from the experimentally tested vocabulary of 1,000 words: (1) 80 "index words" which are the initial elements in the combination call signs and (2) 500 highly intelligible "base words" to serve as the second element in the combination call signs.

By combining these index and base words, it is possible to obtain up to 40,000 separate highly intelligible two-word phrases. From these, the 10,000 combinations which are most satisfactory can be chosen for use as tactical call signs.

Such a list of combination call signs has the disadvantage that each signal is three or more syllables in length. Also, some of the phrases border on the nonsensical. It is doubted, however, that the latter is a real objection to their use. Indeed, in view of the American soldier's irrepressible love for the sassy and ridiculous, the sometimes unusual and amusing products of random combination may be more of a stimulus than a detriment to morale. Few enlisted men fail to derive enjoyment from having their telephone units identified by such names

as "Noah's Schooner," "Joshua's Woodpecker," or "Cornell Apple-jack." The large stock of words available makes it easily possible to eliminate any truly objectionable combinations from the final list of 10,000 call signs. There seems to be no way, however, in which anything else but word combinations can yield the necessary number of signals.

Chapter 7

INTELLIGIBILITY OF SPEECH: EFFECTS OF DISTORTION

VOICE COMMUNICATION equipment used by the Armed Services was improved markedly during the course of World War II, especially as regards the reduction of frequency distortion. Headphones providing a uniform frequency response up to 4,000 c replaced resonant "peaked" phones. The frequency-response characteristics of microphones, interphones, and radios were also improved.

Although the problem of frequency distortion was more vigorously attacked, amplitude distortion was hardly less prevalent in Service equipment. Amplitude distortion is especially marked in military communication because the din of battle and the noise of fighting machines require unusually intense speech. Loud speech drives the communication circuits harder than would conversational speech, and amplitude distortion often results.

The question inevitably arises, therefore, as to what kind and how much distortion can be tolerated without seriously interfering with the intelligibility of speech. The intelligibility of speech transmitted by resonant systems and the effects of amplitude distortion (overloading) were first studied at the Bell Telephone Laboratories,[1] and the purpose of the experiments discussed herein was to supplement and extend these early observations.

7.1 AMPLITUDE DISTORTION[6]

7.1.1 Types of Amplitude Distortion

Amplitude (or nonlinear) distortion results whenever a signal is passed through a nonlinear circuit. For purposes of rough classification, such circuits might be thought of as falling into three categories:

1. Circuits which transmit low-amplitude parts of the wave more efficiently than high-amplitude parts. This is by far the most frequent type of distortion.

2. Circuits which transmit high-amplitude parts of the wave more efficiently than low-amplitude parts.

3. Circuits which transmit parts of the wave on one side of the time axis more efficiently than parts of the wave on the other side of the time axis.

To investigate experimentally the effect of these types of distortion on the intelligibility of speech, it was decided to employ representatives of each of these types of amplitude distortion. Consequently, vacuum-tube circuits were arranged to simulate various amounts of amplitude distortion introduced by:

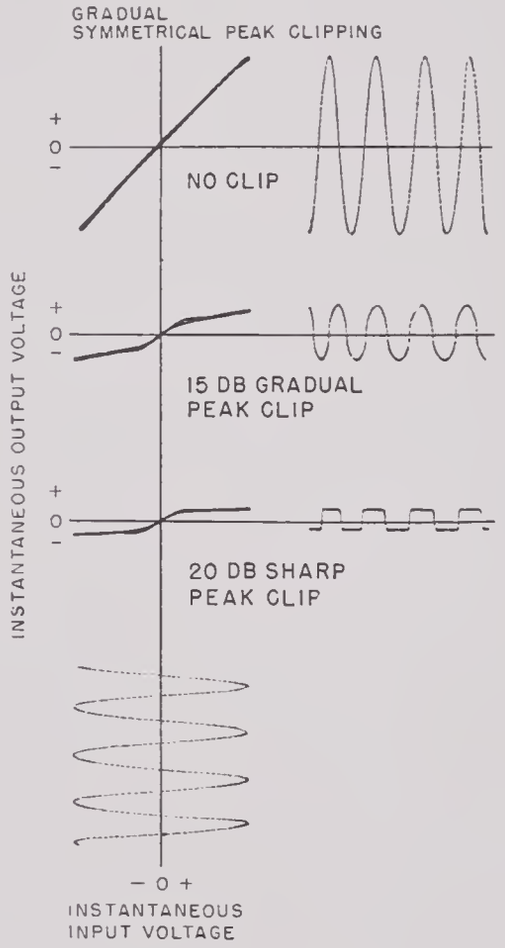


FIGURE 1. Gradual, symmetrical peak clipper. The oscillograms superimposed on the coordinates show instantaneous input-output characteristics of a nonlinear circuit which could be adjusted for linear response, for gradual peak clipping, or for sharp peak clipping.

1a. A push-pull amplifier with a gradual overload characteristic. For convenience, this is referred to as gradual, symmetrical peak clipping (see Figure 1).

1b. A push-pull amplifier which overloads abruptly as the input signal is increased past a critical point. This type of distortion is referred to as sharp, symmetrical peak clipping (see Figure 2).

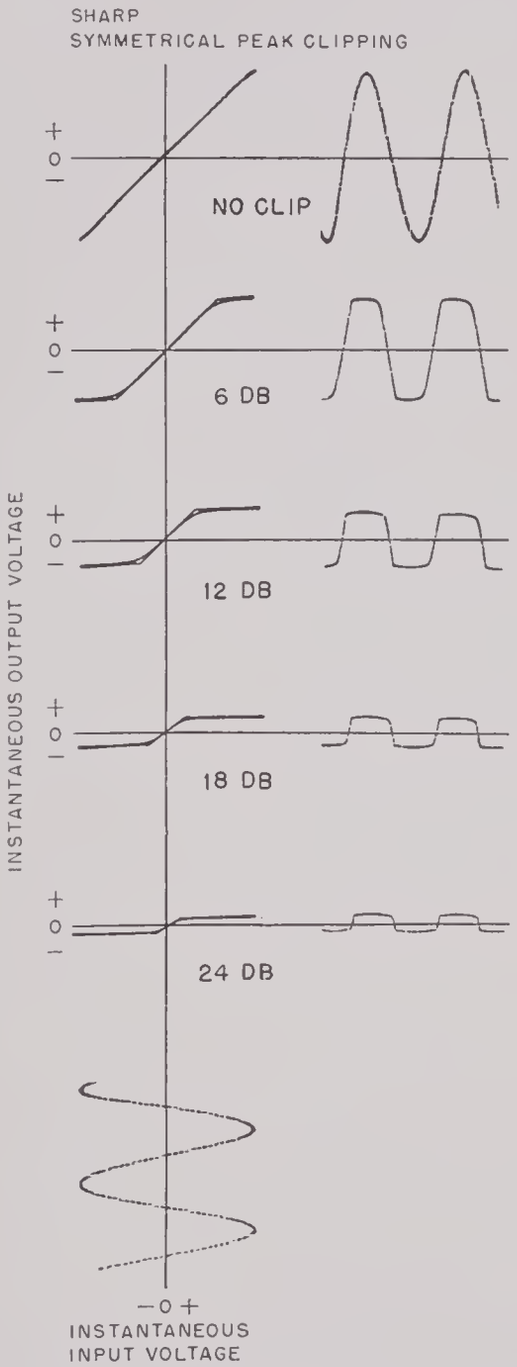


FIGURE 2. Symmetrical peak clipper. The oscillograms show input-output characteristics and wave forms for the sharp symmetrical peak clipping circuit.

1c. An unbalanced amplifier which overloads abruptly on swings in one direction only. This is called a sharp, asymmetrical peak clipping (see Figure 3).

2. A misaligned Class B amplifier which discriminates against weak signals. Such distortion is called center clipping (see Figure 4).

3. A misaligned Class B amplifier which amplifies positive voltage swings less than negative voltage swings. This is called "linear rectification" (see Figure 5).

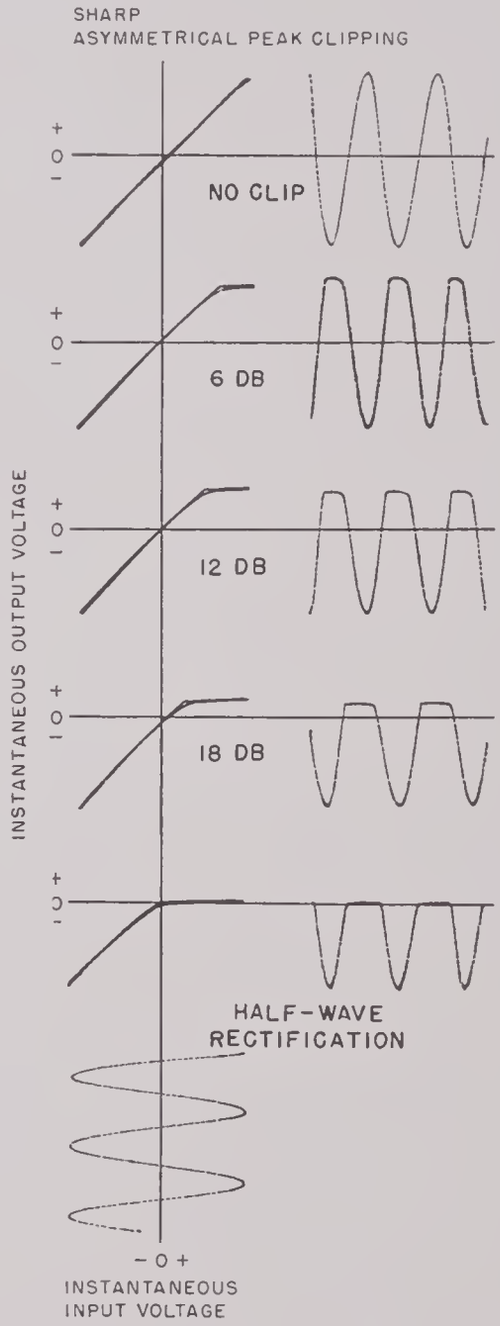


FIGURE 3. Asymmetrical peak clipper. In these oscillograms clipping occurs on only one side of the time axis.

The nonlinearity of such circuits is best described by the input-output characteristics. Two input-output characteristics can be distinguished: (1) The functional relation between

the *effective* (rms) values of input and output voltages, or (2) the functional relation between the *instantaneous* values of input and output voltages. In some circuits, such as "compressing" or "expanding" amplifiers, the effective characteristic is nonlinear, but the instantaneous characteristic is essentially linear. For the circuits listed above, however, both the effective and the instantaneous input-output characteristics are nonlinear. The instantaneous input-output characteristics of these circuits are shown in Figures 1 through 5. The input waves at the bottom, reflected in the input-output characteristics, result in the output waves at the right.

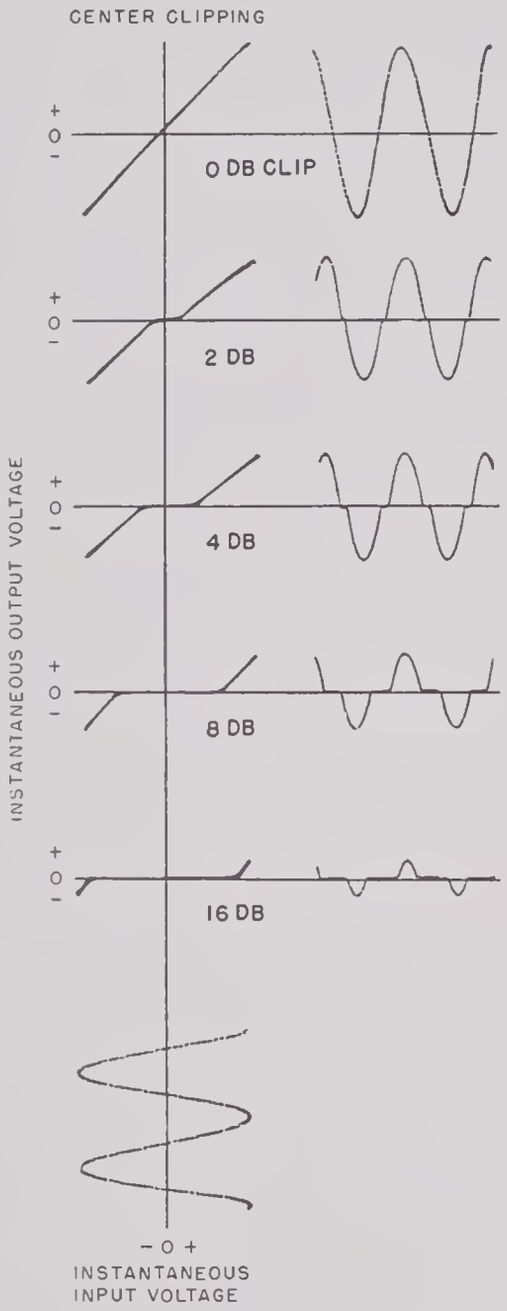


FIGURE 4. Center clipping. The center part of the wave, nearest the time axis, is eliminated and only the peaks of the wave are passed.

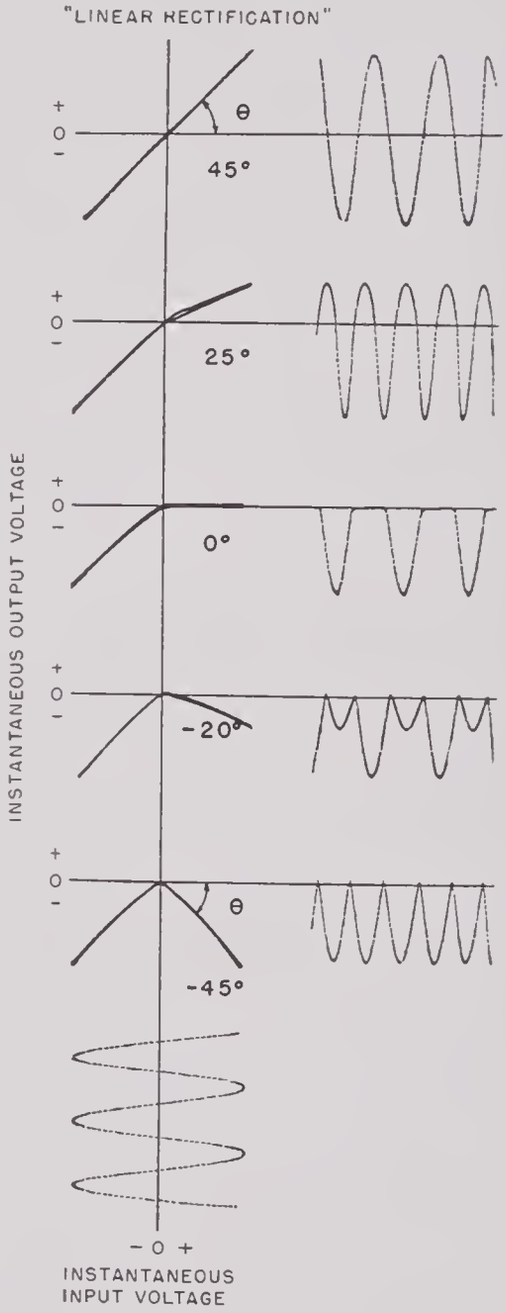


FIGURE 5. "Linear rectification." The distortion depends upon the angle θ of the right-hand leg of the input-output characteristic. The third characteristic represents half-wave rectification.

7.1.2 Measurements of Amplitude Distortion

A particular nonlinear circuit does not necessarily produce similar distortions in dissimilar signals. Figure 6 illustrates this point with

three cases. It is evident from the illustration that the output signal which results when a signal is passed through a nonlinear circuit is determined both by the characteristics of the signal and by the characteristics of the circuit.

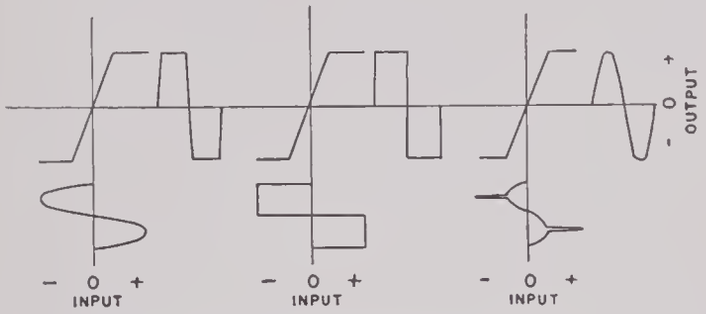


FIGURE 6. Illustrating how the same input-output characteristic may affect different wave forms.

For this reason it is desirable to measure the distortion of the circuit in question with a signal of the type which the circuit is normally called upon to handle.

Because speech is too complex a signal for

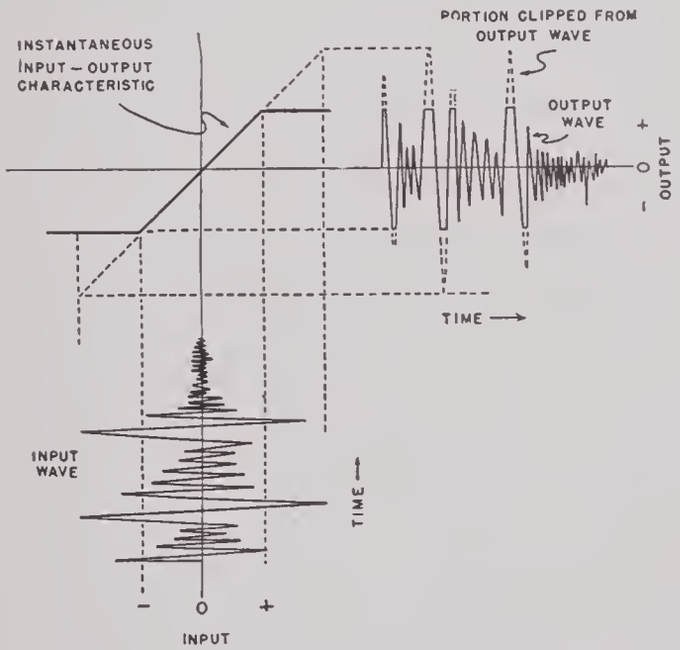


FIGURE 7. Showing the effect of a clipping circuit on a speech wave. The distortion produced can be expressed in terms of the proportional reduction in the amplitude.

simple analytical procedures, however, it is not generally practicable to measure distortion with speech as the test signal. The simplest procedure for handling this problem is to specify the instantaneous input-output characteristics of the distorting circuit and the intensity of the speech signal at the input to the circuit. For indicating distortion, the most appropriate measure of speech intensity is the average peak amplitude of the items in the speech sample. Thus specified, the distortion of the speech wave can then be described verbally as, for example, "sharp, symmetrical peak clipping at one-half peak amplitude," or "6-db sharp, symmetrical peak clipping." Figure 7 illustrates the meaning of these statements.

In distorting a sinusoidal signal, a nonlinear circuit has the effect of producing other frequencies which are harmonics of the input frequency. The power (or voltage) in these harmonics, when stated as a percentage of the power (or voltage) in the fundamental or in the fundamental plus harmonics, can be used as an expression of the degree of distortion. Although the resemblance between speech and sinusoidal waves is remote, several such expressions are in common use and lead to the definition of the term *per cent harmonic distortion* [HD].

Inasmuch as speech is itself comprised of complex waves, complex signals are of interest in expressing distortion. In transmitting a complex wave, a nonlinear circuit may give rise to modulation products or "combination frequencies" equal to sums of or differences between component frequencies of the input signal or their harmonics. An input signal consisting of two frequency components equally intense and harmonically unrelated can, therefore, be used to measure so-called *per cent combination-tone distortion* [CTD]. For this expression of distortion, the harmonics of the two input frequencies are not included as combination tones.

It is especially important that HD or CTD introduced by speech-transmission circuits be measured with test signals of the same peak amplitude as the speech they are normally called upon to pass. If the speech is equated to a sine wave on the basis of effective or average voltage, the high peaks of the speech wave may be distorted by a circuit that would pass sine waves unimpaired.

In Figure 8 typical expressions of HD and CTD are compared for two types of clipping. In the following sections distortion is discussed

principally in terms which indicate the amount of distortion with direct reference to the characteristics of the nonlinear circuit. In addition, however, results have been plotted in terms of

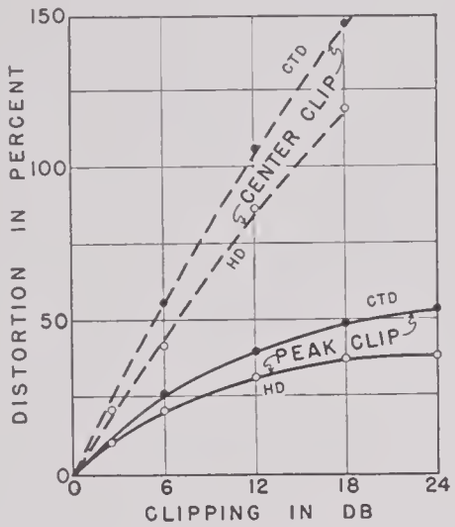


FIGURE 8. Harmonic and combination-tone distortions are shown as functions of two types of clipping.

HD and CTD. Of these two expressions, the latter seems to bear a somewhat more direct relation to the effect of distortion on intelligibility.

7.1.3 Effects of Amplitude Distortion on Intelligibility

The effect of each type of distortion on the intelligibility of speech was determined by means of word articulation tests (see Chapter 5). A nonlinear circuit was introduced into an otherwise linear, wide-band communication system, and the efficiency of communication over the system was measured. Tests were conducted in quiet and with several types and levels of noise.

PEAK CLIPPING

Results with peak-clipping circuits confirm the tests at the Bell Telephone Laboratories in indicating that severe overload distortion can be tolerated without appreciable impairment of intelligibility. The amount which can be tolerated depends upon noise conditions and upon the intensity of the speech signal with respect to the listener's threshold of hearing. Under any but the most extreme circumstances, however, the speech wave (as viewed on an oscilloscope) can be stripped down to one-quarter amplitude (12 db of sharp, symmetrical peak clipping) without any noticeable effect upon the intelligibility of speech. As a matter of fact, the intelligibility of speech is not lost entirely even when the speech wave is clipped and reamplified in such a way as to produce nothing but a succession of "square" waves. Expressed in decibels, this approximates infinite peak clipping.

A comparison of the intelligibility of peak-clipped with undistorted speech must, however, specify the methods of measuring the two signals. If the clipped and unclipped signals are compared for intelligibility at equal peak voltages, the clipped signal is found to be more intelligible. When they are compared at the same effective or average voltage levels, the intelligibility is approximately the same. And if the two signals are equated on the basis of the voltage which the clipped signal would have if it were not clipped, the unclipped speech is more intelligible.

For example, suppose that a typical speech sample is used to compare 0-db with 24-db peak clipping. With previously undistorted speech, the peak voltage (read on a cathode-ray oscilloscope) will be about 12 db higher than the average voltage (read on a VU meter) for individual words. Peak clipping of 24 db (assuming no other voltage gain or loss in the nonlinear circuit) lowers the peak voltage 24 db and also lowers the average voltage approximately 14 db. The output signal now has a peak voltage only 2 db higher than its average voltage.

If this signal is now compared with the unclipped signal for intelligibility, the experimental finding is that the clipped speech must be amplified approximately 14 db in order to give the same articulation score, i.e., when the average voltages of clipped and unclipped speech waves are equal, their intelligibility is approximately the same (up to at least 24-db peak clipping). Thus, 14 db are "lost" in the sense that 14 db more amplification must be provided. Given the same average voltages, however, practically nothing is lost or gained

by clipping. But in terms of peak voltages, the unclipped speech will give the same articulation score with a peak voltage 10 db lower than the peaks of unclipped speech, so 10 db have been "gained."

Thus, it is seen that the effects of peak clipping may prove a handicap in some situations, but in others it may become a benefit. If the gain of the nonlinear circuit (defined as the gain for unclipped signals) is constant, but the clipping level is variable, clipping will lower intelligibility. When the clipping level is constant, however, and the gain is variable, clipping improves intelligibility. It is fortunate that in the majority of communication situations the latter is more nearly the case than the former.

Thus far we have considered only the case of peak-clipping speech heard in quiet. The effect of adding noise depends upon whether the speech and noise are mixed before or after the distortion is introduced. If the noise is added *after* the speech has been distorted, the effect is to raise the listener's threshold of hearing (masking) but otherwise the statements above are still applicable. The only exception occurs when the noise is so intense that the speech intensity reaches the ear's threshold of feeling. Under this condition, clipping the speech peaks acts to protect the ear, and greater intelligibility can be obtained.

Data supporting these general conclusions are shown in Figures 9, 10, and 11. These three figures are drawn from the same data and represent the three ways of expressing the effects of peak clipping discussed above. The random-noise interference had a spectrum of constant energy per octave band, and the two levels of noise used were equivalent to 90 and 120 db re 0.0002 dyne/cm^2 measured in a 6-cu cm coupler. In Figure 9 the data are plotted in terms of the gain in the speech channel and demonstrate that more amplification is required for peak-clipped speech to attain the same articulation score. Figure 10 presents these data in terms of the average level of the speech signal received at the listener's headphones (as indicated by a VU meter) and shows that articulation is relatively independent of peak clipping when the same average level is maintained. Figure 11 shows these data when the speech level is taken as the peak amplitude of the speech signal which reaches the listener's ears. These curves demonstrate that when peak values are the limiting factor the intelligibility of speech is improved by clipping the speech peaks as much as 24 db.

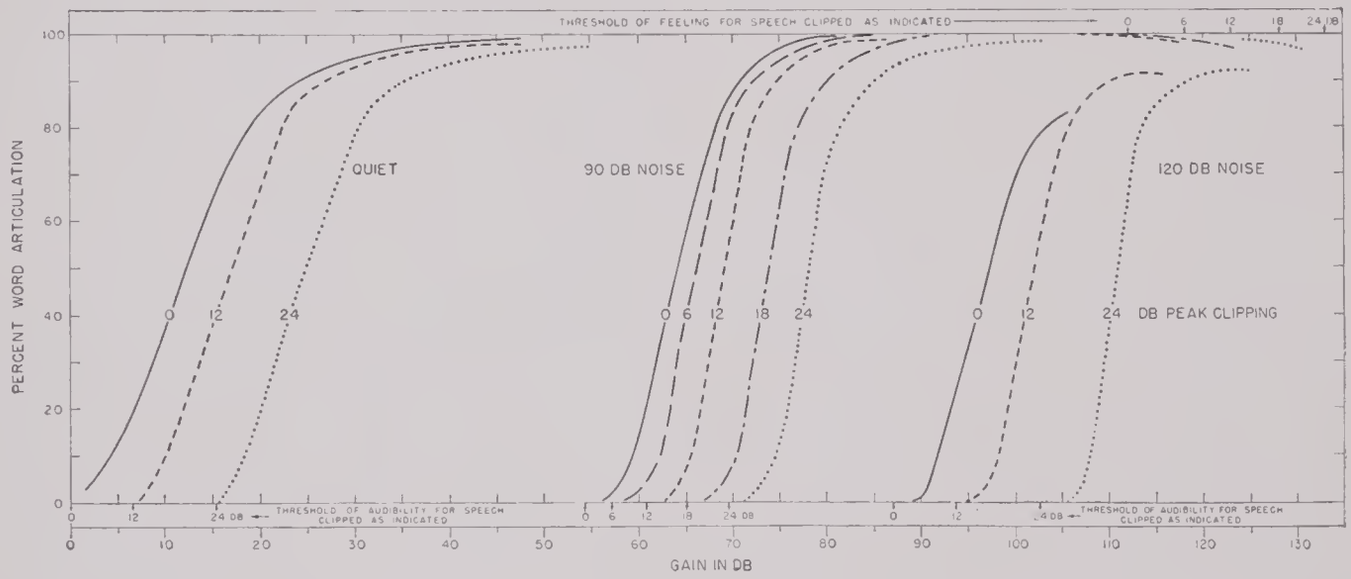


FIGURE 9. Functions relating articulation to gain with peak clipping and background noise as parameters.

When the speech and noise are mixed *before* the peak clipping occurs, then both speech and noise may be distorted by the nonlinear circuit. Cross modulation between the two signals would be expected to result in more severe impairment of intelligibility than was found in the experiment above.

One practical situation occurs when an unshielded microphone is used in intense ambient noise. The detrimental effects of peak clipping are slightly more pronounced when the articulation tests are conducted with the announcer and listeners in an intense ambient noise field. This result was obtained for various ambient noises and types of amplitude distortion. Similarly, the detrimental effect increases as the amount of noise picked up by the microphone increases. Speech from a "close-talking" microphone (MC-253, magnetic), which picks up relatively little ambient noise, suffers less impairment from peak clipping than does speech from a standard condenser microphone (WE 640-A) which has no provision for noise exclusion. This result is shown in Figure 12. Tests with a throat microphone (T-30-P) also revealed little impairment in intelligibility with peak clipping. In spite of the poor quality of

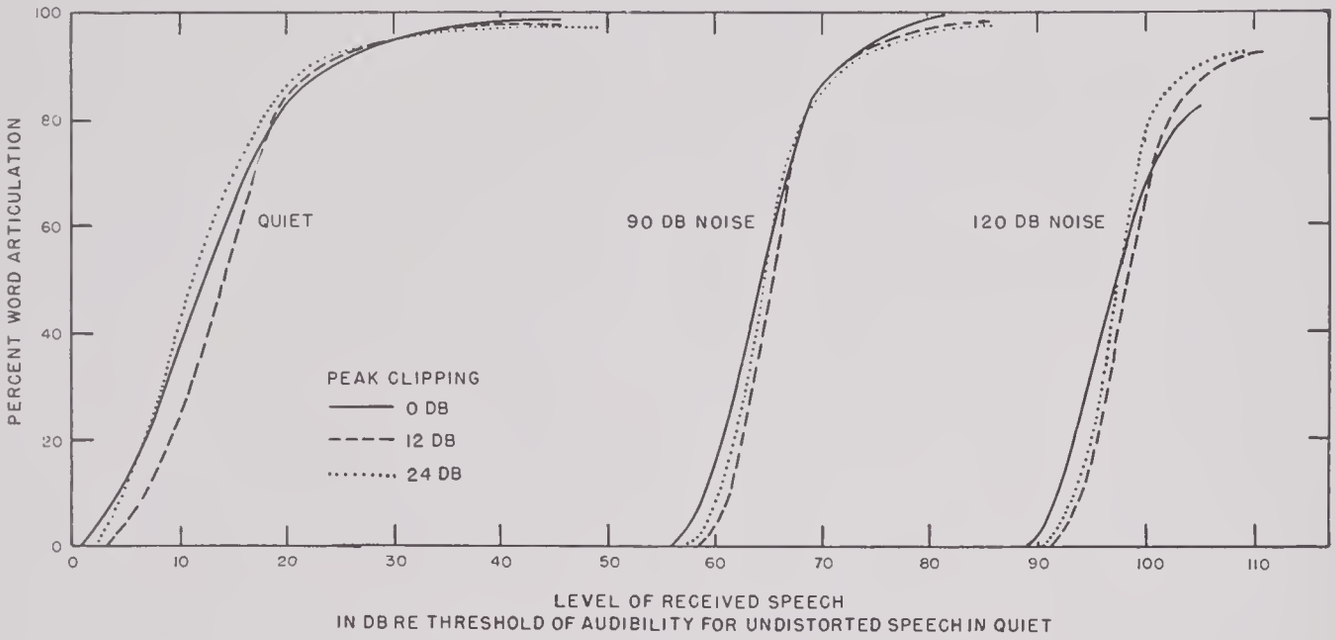


FIGURE 10. Functions relating articulation to level of received speech with peak clipping and background noise as parameters.

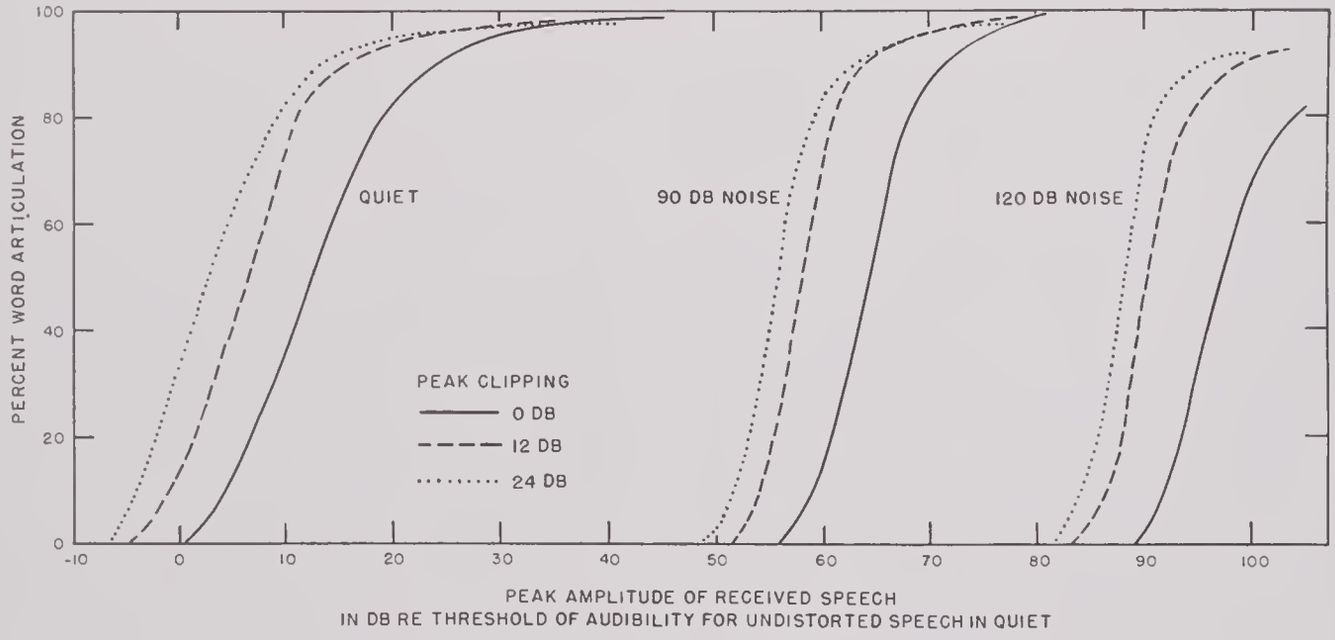


FIGURE 11. Functions relating articulation to peak amplitude of received speech with peak clipping and background noise as parameters.

throat-microphone speech, the throat microphone picks up relatively little noise to be intermodulated with the speech, and thus proves relatively resistant to the effects of such distortion.

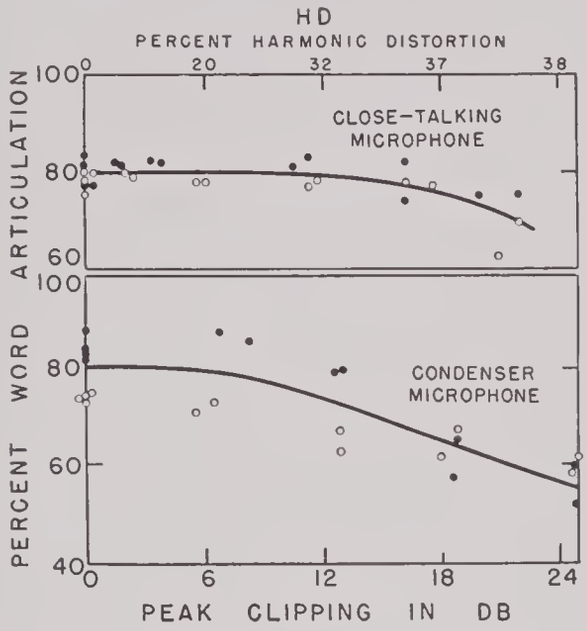


FIGURE 12. Articulation as a function of peak clipping. Talker and listeners were located in random noise field of 110 db. Filled and unfilled circles distinguish two talkers.

More complete data relative to this problem were obtained by the check-list testing method. The results are shown in Figure 13. The amount of peak clipping is here specified by measurements made with speech alone. The spectrum of the unclipped noise was flat by octaves. The noise level when not clipped corresponded to 100 db re 0.0002 dyne/cm^2 in a 6-cu cm coupler. The intensity of this noise was held constant at the input terminals of the clipping circuit. Since speech and noise together reach peak amplitudes higher than speech alone, it is necessary to distinguish between "no clipping" of the combined signals and clipping at the average peak amplitude of the speech alone (0-db clipping).

The several amounts of clipping yield a family of curves separated by only about 5 db. Comparison of Figure 13 with Figure 9 shows that peak clipping causes less impairment of intelligibility when noise is mixed with the speech before distortion. This is due to the fact that clipping the peaks reduces the average noise voltage about as much as it reduces the average speech voltage. Consequently the signal-to-noise ratio is decreased less when the combined signal is clipped than it is when the speech alone is clipped and then mixed with the noise.

This comparison does not, however, present the complete picture. When speech was clipped before being mixed with noise, it was seen that the loss in intelligibility could be made up by additional amplification. When the speech and noise are mixed before clipping occurs, however, any subsequent amplification will change both speech and noise, thus leaving the speech-

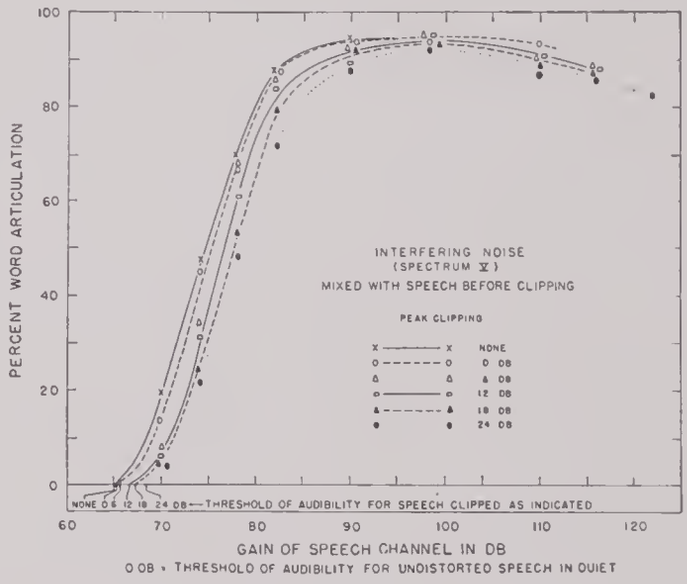


FIGURE 13. The curves show the relation between articulation and the gain of the speech channel when a constant amount of noise was mixed with the speech before the signal was clipped. The amount of clipping indicated for each curve was determined in the absence of noise.

to-noise ratio unaffected. For this reason it must be concluded that, whenever possible, the signal should be kept free from noise until it has been clipped.

A second practical case where speech and noise may be mixed before they are peak clipped occurs in radio-telephone communications (see Chapter 13). So-called "noise-peak-limiter" circuits are used in radio receivers to minimize pulse-type interference and to improve intelligibility. Some of these limiters are essentially peak clippers. The results just described show that impairment results from cliping a mixed signal if the noise has strong

low-frequency components and a relatively low peak factor. It is of interest, therefore, to examine results for intermittent, high-peaked interference of the type frequently encountered in radio communication.

Tests were conducted as described above, but with audio static as interference. Because the static spikes were of extremely high amplitude, it was necessary to clip at amplitude levels corresponding to twice and four times the average peak amplitude of the test words. For convenience, these additional clipping conditions are designated as —6 db and —12 db, respectively.

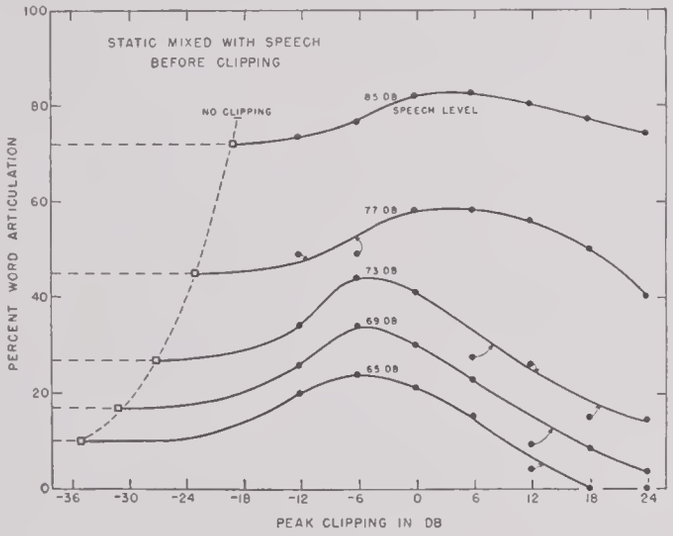


FIGURE 14. Effects of peak clipping when static passes with speech through nonlinear circuit. The levels of the speech signal are expressed in decibels above the threshold for undistorted speech in quiet. The negative values of peak clipping affected the static but not the speech.

In Figure 14 the articulation results are plotted against peak clipping. A curve is shown for each level of received speech at which the tests were conducted. Since the static was maintained at a constant level, the increasing levels of received speech correspond to increasing speech-to-static ratios.

With the particular type and level of static used, the optimal amounts of clipping lie between 6 db (for higher levels) and —6 db (for lower levels). The amount of improvement in intelligibility under the most effective conditions is between 10 and 20 articulation points. These data indicate, therefore, that peak clipping at the proper amplitude level can improve the intelligibility of speech in the presence of static.

Subjective observations of the effects of peak clipping on the quality of the speech indicate that the quality is impaired surprisingly little. On the contrary, with 12-db peak clipping it sounds as though the speaker were enunciating with special care. Although small amounts of peak clipping are not objectionable, tests with speech signals have shown that the point at which clipping begins can be detected almost as accurately by ear as by examination of the trace on a cathode-ray oscilloscope. The sensitivity of the ear to these slight changes in harmonic content is surprisingly acute (see Section 3.5.1). The visual determination is to be preferred, however, because of its greater reliability.

Speech is still quite intelligible with 24-db peak clipping, but sounds unnatural and "grainy." In general, the intensity of the peak-clipped speech is monotonously uniform, and this is attributable to the fact that the clipper circuit reduces the level of the more intense words but leaves the weaker words relatively unaffected.

CENTER CLIPPING

The extreme tolerance of the ear for peak clipping is matched by an equally marked intolerance for center clipping. The results of center-clipping tests are shown in Figure 15. The comparable curve for peak clipping is replotted from Figure 12. More than 1-db center clipping severely reduces the articulation scores.

An explanation of the highly dissimilar effects of peak and center clipping can be given in terms of certain characteristics of a typical speech wave. The intelligibility of speech depends primarily upon the correct identification of consonant sounds, while the recognition of vowels is much less critical. The nature of speech is such, however, that the less important vowel sounds are about 10 to 15 db more intense than the very important consonant sounds. Consequently, when speech is peak clipped it is primarily the vowels which are affected; the consonants are increased in relative amplitude, and the speech remains in-

telligible. With center clipping, on the other hand, the weak but important consonant sounds are proportionately more affected than the vowels, and intelligibility is severely impaired.

Center clipping produces subjective effects much more marked than those of the types of

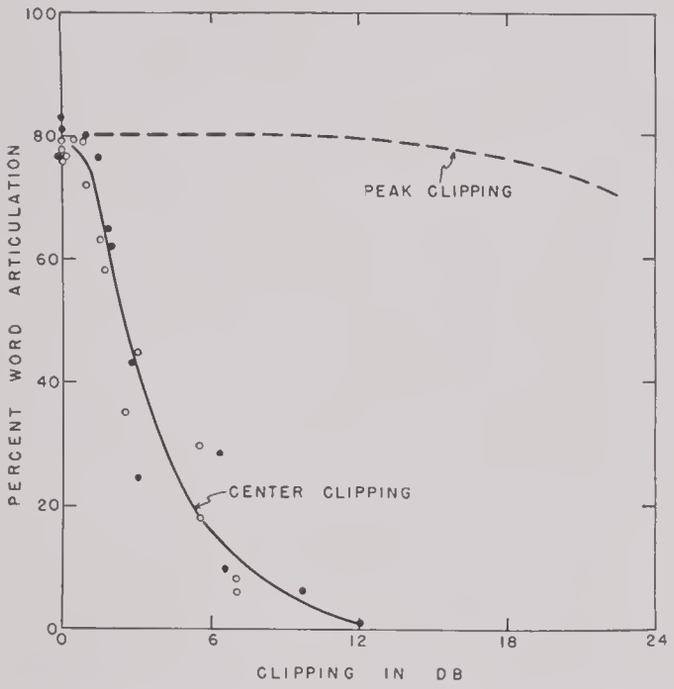


FIGURE 15. Comparison of effects of peak clipping and center clipping. Talker and listeners were located in random noise field of 110 db. Filled and unfilled circles distinguish two talkers.

distortion previously described. Even slight amounts of center clipping give speech a peculiar "lisplike" quality. It has also been noticed that center-clipped speech sounds somewhat like the speech transduced by a throat microphone. Increasing amounts of center clipping produce, first, a "coarse," "granular" sound, followed by the effect of intermittent breaks or shorts in the circuit. Severe center clipping causes speech to sound rather like radio static.

Inasmuch as center clipping eliminates the weak sounds, it is possible that use could be made of center-clipping circuits in training speakers to talk effectively in noise. A person listening to himself through a center-clipping circuit has a natural tendency to emphasize the weak consonants which, otherwise, he would not hear on his side-tone circuit. The circuit might also find application in connection with "speech interviews" in which it is desired to rate individuals as to their probable effectiveness in talking against severe interference.

Linear Rectification

The relatively small amounts of "linear rectification" which might occur in a poorly balanced Class B amplifier were found to cause no serious impairment of intelligibility. Articulation scores, as shown in Figure 16, were down 10 to 20 per cent, however, when the speech

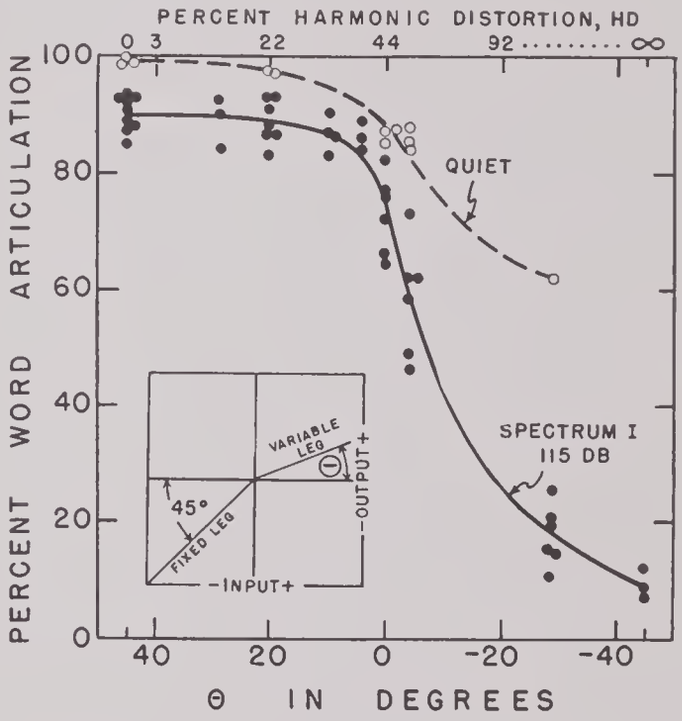


FIGURE 16. Articulation as a function of "linear rectification."

wave was half-wave rectified. Full-wave rectification was extremely detrimental to speech quality and reduced intelligibility severely.

7.1.4 Applications of Peak Clipping

It has been shown that peak clipping does not seriously impair the intelligibility of speech. In some cases, where the peak amplitude of the wave is maintained at a constant level, irrespective of clipping, it is found that peak clipping may actually improve the communication situation. At least three such cases have presented themselves, and research has been done to determine the advisability of deliberate peak clipping.

One application occurs in the use of hearing

aids. When the speech is amplified to the point at which it can be heard by people with severe deafness, the peaks of the wave may reach the feeling threshold of the ear. The possibility of using peak clipping in hearing aids to protect the ear at high intensity levels is considered in detail in Chapter 15.

A second use for peak clipping occurs in amplitude modulation [AM] radio transmission. In this case the higher speech peaks may overmodulate the transmitter, produce interchannel interference, and overwork the modulator unit. Peak clipping makes it possible to maintain 100 per cent modulation at all times.

The third case also occurs in the realm of radio telephony. As shown in Figure 14, peak clipping of the proper amount may improve the intelligibility of speech heard in interfering static. This application is presented in Chapter 13, where the use of noise limiters in radio receivers is discussed.

The general conclusion must be, therefore, that amplitude distortion is not always undesirable. If the interest lies in intelligibility rather than high quality, certain types of amplitude distortion may actually be advisable. Peak clipping, which provides greater intelligibility per unit peak amplitude, seems to offer a number of possibilities for improving communication efficiency.

7.2 FREQUENCY DISTORTION

Frequency distortion occurs whenever a transmission system responds unequally to sinusoidal signals of different frequencies. It is obvious, of course, that lack of uniform response for frequencies above or below the audible range cannot be detected by the ear, and consequently the term "frequency distortion" as used herein refers to the response of the system for audio frequencies.

7.2.1 Concept of Orthotelephonic Gain

Meaningful comparisons between two systems can be most readily obtained by defining a suitable reference system. A reference system which uses real talkers and listeners and which is not too difficult to reproduce in the laboratory has been defined at the Bell Telephone Laboratories.[4] It is called orthotelephonic reference system, and it consists of a "normal" talker and listener facing each other in open air at a distance 1 m apart in a quiet, nonreverberant room.

The overall frequency response of a system under test is then expressed relative to that of the orthotelephonic system, and the response so obtained is called the orthotelephonic gain of the system. The basic notion is that the orthotelephonic gain of a system at a given frequency shall be expressed as the ratio of the level produced at the listener's ear by the system under test to that at the listener's ear when speech is transmitted from talker to listener over a 1-m path in free air. By definition, then, speech will sound the same over a system having an orthotelephonic gain of 0 db at all frequencies as it will when heard over an air path 1 m long.

In practice it is often more convenient to obtain the overall response of the system to be tested in three separate steps (microphone, earphone, and amplifier). More often than not, human talkers and listeners are replaced by artificial voices and ears. Although approximate only, such response characteristics are useful. For a more complete discussion, see Chapters 9 and 10.

7.2.2 Types of Frequency Distortion

As a matter of convenience, circuits with nonuniform frequency responses can be classified into four types:

1. Circuits which pass high frequencies more efficiently than low frequencies.
2. Circuits which pass low frequencies more efficiently than high frequencies.
3. Circuits which pass intermediate frequencies more efficiently than high and low frequencies. This is the most common type of frequency distortion.
4. Circuits which pass high and low frequencies more efficiently than the intermediate frequencies.

These various types of frequency distortion are conveniently produced for study by high-pass, low-pass, band-pass, and band-elimination filters.

In most cases an adequate statement of the type of distortion can be made in terms of cutoff frequencies and/or the slope of the response characteristic over the frequency ranges of nonuniform amplification. These two specifications are adequate, however, only in idealized cases. When the single-frequency response characteristic is complex and irregular, the only adequate definition of the frequency distortion is the graphical representation of the characteristic.

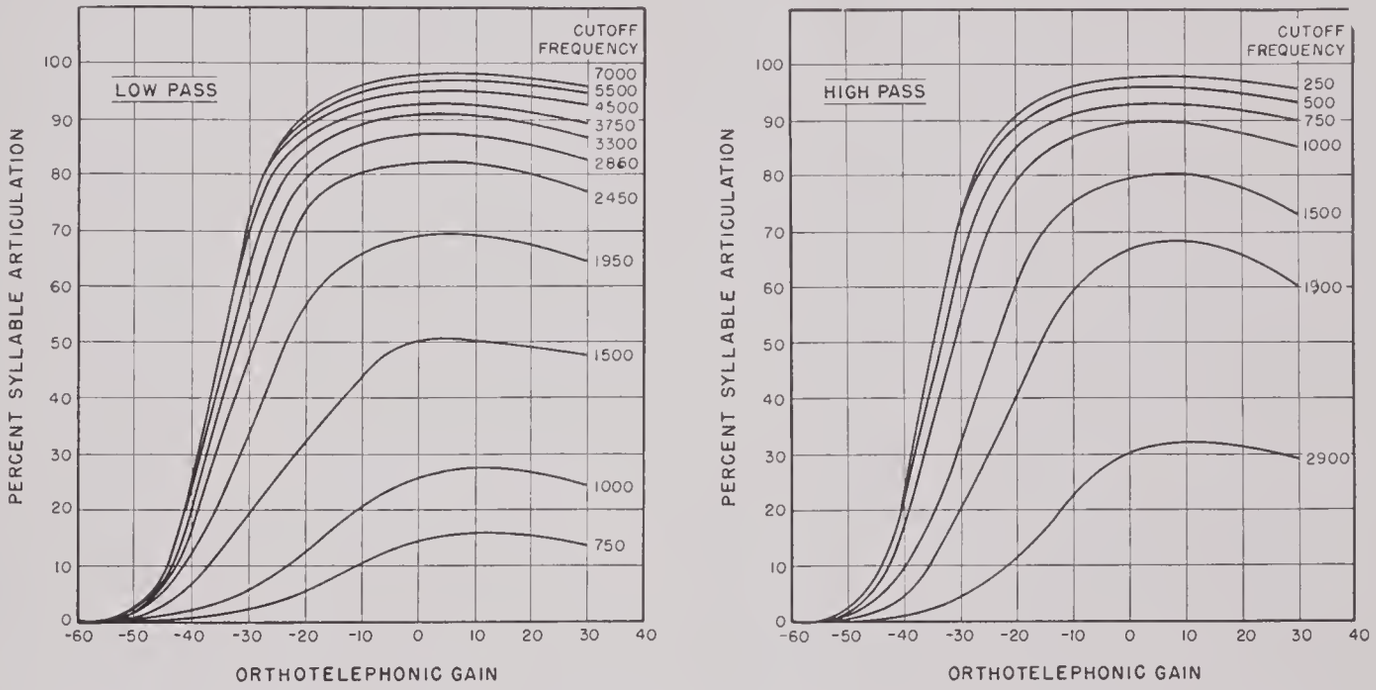


FIGURE 17. Functions relating syllable articulation to orthotelephonic gain with the cutoff frequency of filters as parameter.

7.2.3 The Effects of High-Pass and Low-Pass Systems

The effects of high-pass and low-pass circuits on the intelligibility of speech have been studied with two kinds of filtering. Data gathered several years ago at the Bell Telephone Laboratories related the articulation score to the cutoff frequency of sharply discriminative filters. The results of a few tests of this type are contained in reference 2. These data are now supplemented by results obtained with circuits having a continuous slope throughout the range of speech frequencies. The articulation score can, in this case, be related to the slope of the "tilted" response characteristic.

Effects of High-Pass and Low-Pass Filters

The relation between articulation scores and frequency distortion can be investigated with highly discriminative high-pass and low-pass filters inserted in a good quality broad-band transmission system. Comprehensive tests of this type made by the Bell Telephone Laboratories[a] are summarized in Figure 17 for high-pass and low-pass filters. These smoothed-gain functions are drawn from composite results for both men's and women's voices. Nonsense syllables were used as test material.

The articulation scores rise rapidly as the orthotelephonic gain of the passed frequencies increases. The optimal scores are obtained for each of the filter conditions at about the same point, roughly + 10-db orthotelephonic gain

[a] These data are as yet unpublished in complete form. They were made available to the Psycho-Acoustic and the Electro-Acoustic Laboratories in order to aid the war effort.

for the particular speaking levels used. If these optimal scores are now plotted against the filter cutoffs, the curves shown in Figure 18 are obtained.

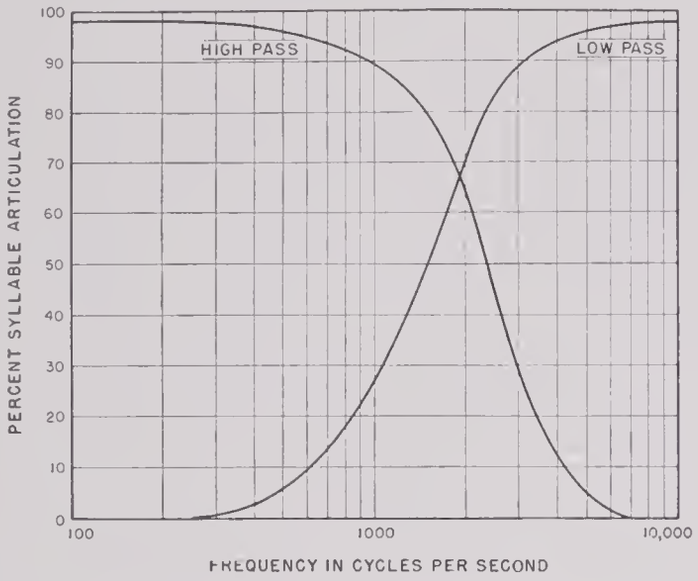


FIGURE 18. Functions relating syllable articulation to the cutoff frequency of high-pass and low-pass filters. Data taken from Figure 17 at +10 db orthotelephonic gain.

It will be seen that the low frequencies of speech contribute very little to the articulation score in spite of the fact that they carry most of the speech power (see Chapter 4). For example, when all the frequency components of speech below 1,000 c were attenuated by a high-pass filter, the speech power was reduced approximately 80 per cent, but the articulation score fell only 10 per cent. Also, it should be noted that, according to these data, the frequencies above 1,900 c make the same contribution to the articulation score as do the frequencies below 1,900 c.

For military purposes it is especially desirable to know the effects of high- and low-frequency cutoffs when the equipment is used in noisy environments. Tests conducted at the Psycho-Acoustic Laboratory related the articulation score to the low-frequency cutoff,[5] and also to the high-frequency cutoff[5a] of several standard military equipments. Results obtained for different low-frequency cutoffs with four microphones under a variety of noise conditions are shown in Figure 19. In all cases the effect of raising the low-frequency cutoff is to lower the intelligibility of speech. The loss is greatest when the noise conditions are least favorable, and in general the rule holds that the more difficult the conditions the wider should be the band of frequencies passed by the interphone. This result is surprising in view of the fact that the noise used in the tests simulated the spectrum of airplane noise and contained a large proportion of the energy in

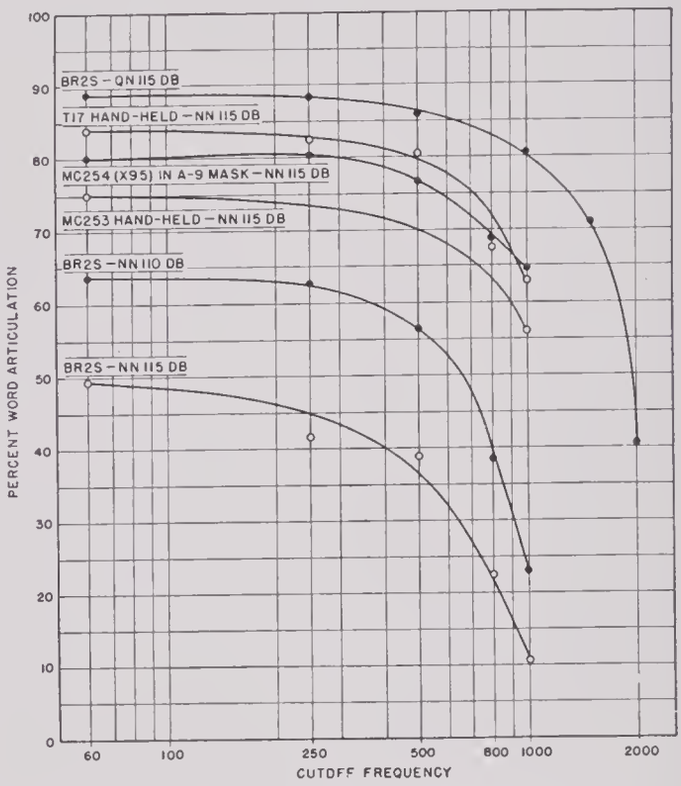


FIGURE 19. The effect on articulation of varying the low-frequency response. Parameters represent different microphones and noise conditions.

frequencies below 250 c. Apparently, for linear systems it is more important to pass the lower speech frequencies than to exclude the lower components of the noise.

Tests were also made to compare a high-frequency cutoff of 3,000 c with one of 4,000 c. Three different microphones and five different noise conditions were used in the tests, and under all conditions the cutoff at 4,000 c gave higher articulation scores than the cutoff at 3,000 c. The advantage ranged from 1 to 7 per cent. In view of this consistent advantage, it was recommended that the earphone be designed to have a flat response as a function of frequency and a cutoff at a frequency not lower than 4,000 c. This recommendation was re-

sponsible for the design of the ANB-H-1 earphone.

Effects of "Tilted" Response Characteristics

A second type of frequency distortion has been studied at the Psycho-Acoustic Laboratory. Filters having gradual, sloping cutoffs were used, and the response over the range of speech frequencies was "tilted" as shown in

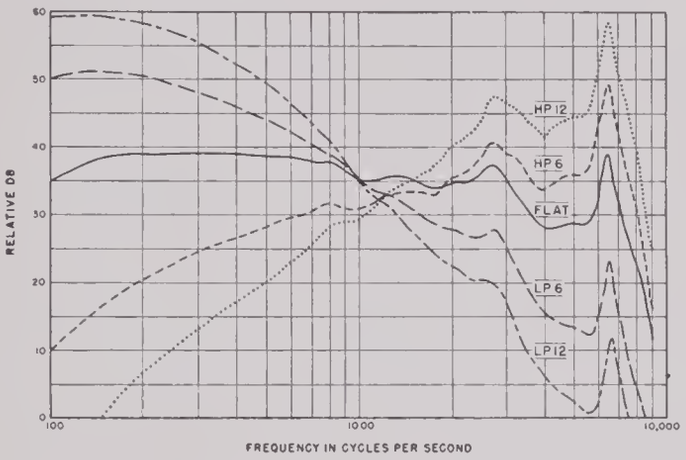


FIGURE 20. Overall frequency-response characteristics for different tilts measured with a 6-cu cm coupler.

Figure 20. The solid lines describe the overall response[b] of the system when no filters were used and is considered "flat" or uniform. The broken lines indicate the response when filters were used to obtain high-pass and low-pass tilts of 6 and 12 db per octave. The relative levels of the tilted response characteristics are adjusted in such a way that all five systems yield the same articulation score (50 per cent word articulation in quiet).

It will be noted that for equal articulation scores the gain of the five systems is approximately the same between 1,000 and 1,500 c. Most of the speech power is carried by frequencies below 1,000 c, however, with the result that the intensity of the speech received via the low-pass systems is necessarily greater (for 50 per cent word articulation) than the intensity of the speech received via the high-pass systems. This fact is revealed by Figure 21, where the intensity of the received speech is indicated both by the peak sound pressure and the average sound-pressure levels [SPL] (measured by a VU meter) necessary to obtain 50 per cent word articulation in quiet. The

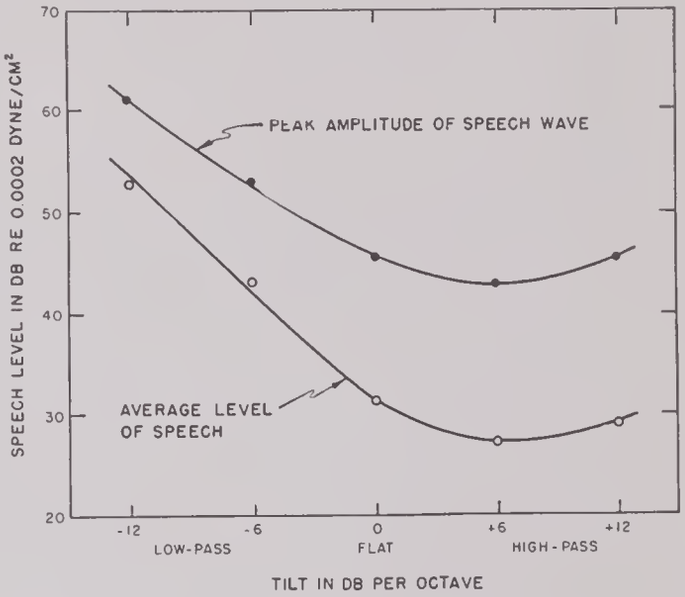


FIGURE 21. Showing the peak and average sound-pressure levels of the received speech necessary in quiet to obtain 50 per cent word articulation with different tilts.

received intensity of the low-pass speech is almost 20 db greater for the same articulation score.

It can also be seen in Figure 21 that the difference between the peak and average levels of the speech (the peak factor) is smallest for the low-pass system, largest for the high-pass. The change in peak factor indicates a change in wave form, and this alteration of the speech wave is illustrated in Figure 22 for high-pass (+) and low-pass (—) systems. These oscillograms of the word "show" also reveal a systematic change in the ratio of the consonant and vowel intensities. With low-pass tilts the consonant-vowel ratio is very low, but with high-pass tilts the consonants are relatively much stronger. This tends to make the speech wave more nearly uniform in level.

When similar articulation data are obtained under noise conditions, however, the results are somewhat modified. In Figure 23 the speech-to-noise ratio (peak voltage of speech to rms voltage of noise) at the threshold of intelligibility is plotted against the tilt of the response characteristic. Comparison of Figures

[b] For purposes of calibration the acoustic output of the headphones was measured with a 6-cu cm coupler (artificial ear).

21 and 23 shows that systems incorporating low-pass tilts provide relatively higher intelligibility in noise than in quiet.

This difference is probably attributable to the difference between the masked and the quiet thresholds of hearing (see Section 3.4.1). In quiet the threshold of hearing is higher for low frequencies, and low-pass systems must therefore provide a higher signal level than high-pass systems. In intense random noise, however, the masked threshold is higher for high frequencies (see Figure 9 in Chapter 3), with the result that low-pass systems are more efficient. Such assumptions depend upon the

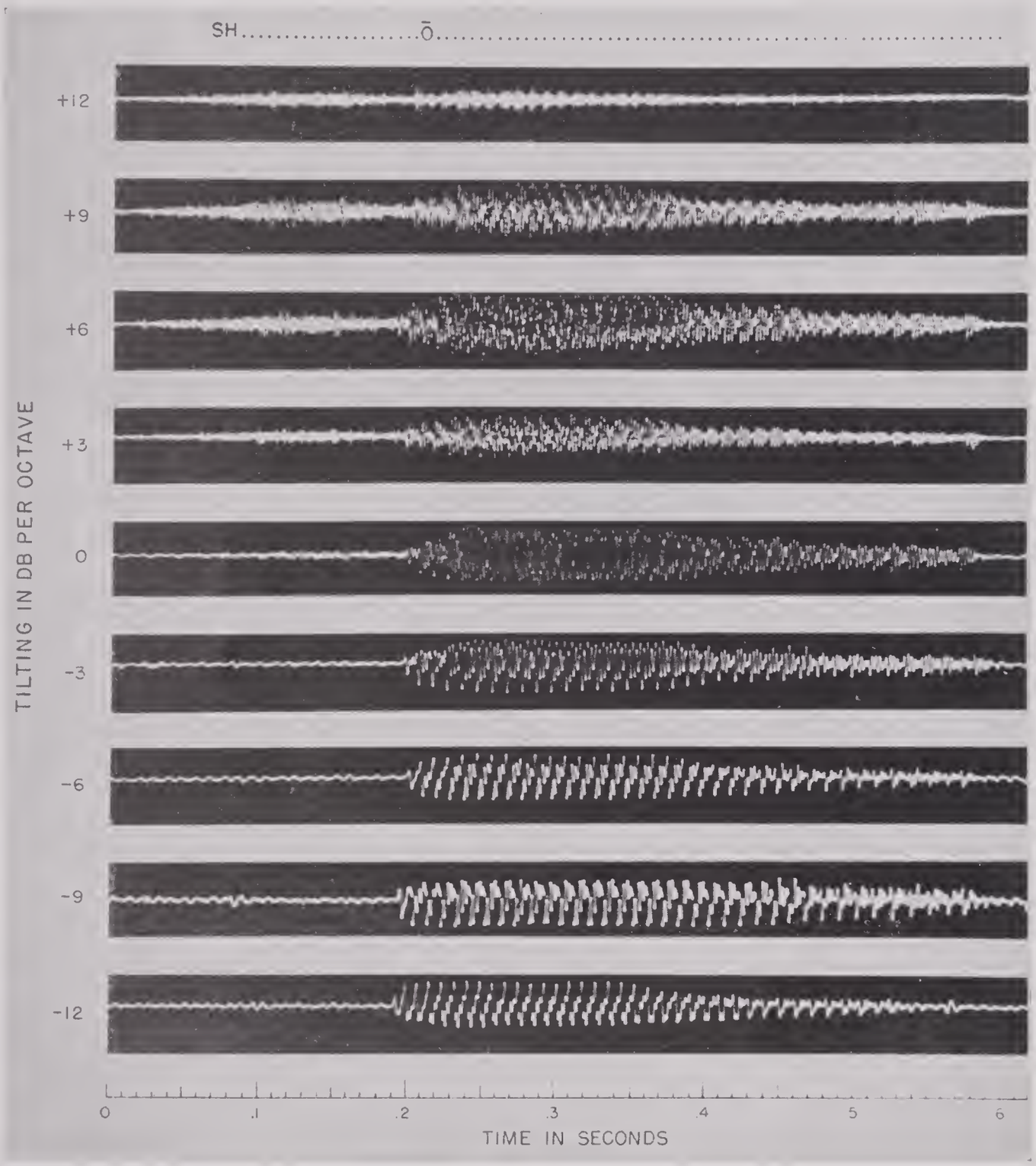


FIGURE 22. Oscillograms of the word "show" after different amounts of tilting.

detectability of tones in quiet and in noise, however, and these do not tell the complete story. The three curves of Figure 23 represent the thresholds of intelligibility, perceptibility, and detectability (see Chapter 5) for continuous speech. Extreme low-pass systems make speech easier to detect in the presence of noise but not easier to understand.

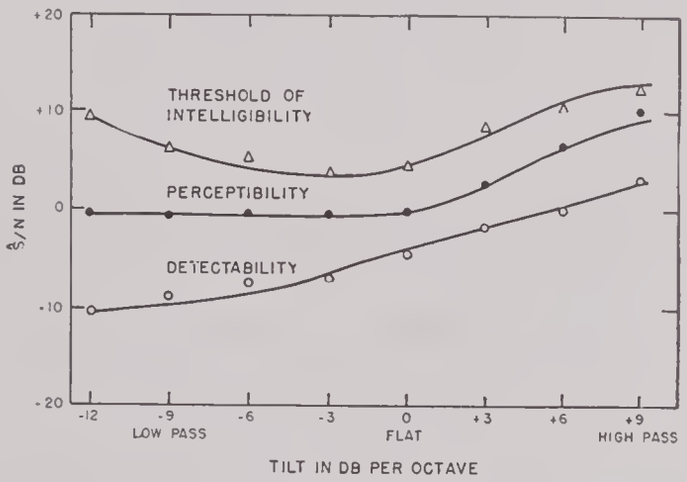


FIGURE 23. Effects of tilting upon the thresholds of intelligibility, perceptibility, and detectability of continuous speech in the presence of random noise.

7.2.4 Effects of Band-Pass Systems

The perception of speech does not utilize all of the frequencies which the ear can detect, and for most speech communication devices it is economical to tailor the frequency response of the system to the necessary speech frequencies rather than to the entire audible range. Engineering considerations often make it desirable to limit the range of frequencies even further. It becomes necessary to know, therefore, what is lost in intelligibility as the band of frequencies is made narrower and narrower, and it is often useful to have at hand the quantitative relations among intelligibility, bandwidth, and level of received speech, evaluated under the acoustic stress of interfering noise.

Effects of Band-Pass Filters

Some of these relations have been determined by articulation tests conducted in noise with communication systems having a variety of band-pass characteristics.[7] The test material (nonsense syllables) was spoken into 12 systems having bandwidths ranging from about one-half octave to the entire range of speech frequencies. The responses of these systems were approximately uniform over the pass band and had sharp cutoffs at either end. The acoustic gain of the pass bands was expressed in terms of the orthotelephonic reference system. Each system was tested over a wide range of speech-to-noise ratios. In all cases the speech-to-noise ratio was varied by altering the gain of the system under test, while the voice level of the talker and the level of the noise in the listener's ears were held constant.

Two spectra of masking noise were used. The noises were intended to bracket the wide range of noise conditions likely to be encountered in military situations. The noise was introduced electrically into the system in a manner designed to simulate both the condition in which only the talker is in an ambient noise and the condition in which only the listener is in the noise.

The results of the syllable articulation tests can be presented in the form of gain functions which express the relation between articulation efficiency and the level of the received speech. Perhaps a more useful form, however, is the equal-articulation contour which shows the relation between level of received speech and bandwidth necessary to maintain a given articulation score.

Illustrative data are presented in Figures 24 through 28. These data were obtained with a noise spectrum containing primarily low-frequency components, and represent the case where the listener is located in a typical ambient noise field (see Chapter 2). The family of gain functions obtained for the group of bands having a center[c] frequency of approximately 1,100 c is shown in Figure 24. Similar families of gain functions were obtained for band-pass systems having center frequencies of 1,500 and 2,000 c.

If a horizontal line is now drawn through such a family of gain functions, the intersec-

[c] The center frequency is defined here as the geometric mean of the two cutoff frequencies.

tions mark off on the abscissa the values of gain required for each of the bands to give the same articulation score. Repeating this process for various articulation levels produces a family

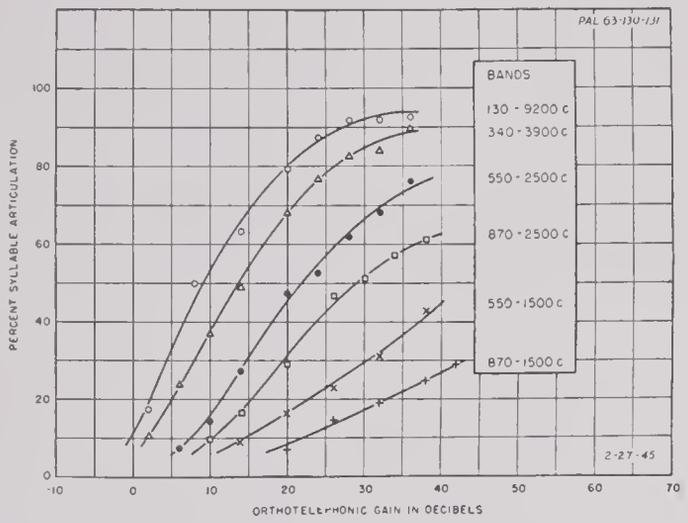


FIGURE 24. Gain functions for band-pass systems in the presence of noise.

of equal-articulation contours. In Figures 25 to 28 are shown four sets of equal-articulation contours drawn from the gain functions of Figure 24 (supplemented by tests with several additional band-pass systems).

These contours are typical of the results obtained with other types of noise and with other center frequencies. Since in each case the family of contours applies only to the specific test conditions, an attempt was made to derive from the data a set of generalized equal-articulation contours applicable to a wider variety of cases. The derivation of these functions involves a certain amount of smoothing and averaging, but in spite of this it is felt that the generalized functions serve a useful purpose in illustrating the general nature of the relations between bandwidth level and articulation.

The generalized equal-articulation contours were all plotted relative to the level of the wide-band system (130 to 9,200 c) required to give an equivalent articulation score. The value of 0-db relative gain was thus assigned arbitrarily to all points of the gain function for the wide band. By this procedure families of contours were constructed for bands with center frequencies at 1,100, 1,500, and 2,000 c for each noise condition tested. Since the resulting sets of contours showed a remarkable similarity, it was possible to represent all of them by the single generalized family of contours shown in Figure 29. This generalized family of contours was constructed with a center frequency of 1,500 c, but to a reasonable approximation the same contours can be used for other center frequencies by shifting the entire family horizontally one way or the other on the semilog plot. The contours probably should not be used for bands having center frequencies outside the range from about 1,000 to 2,000 c.

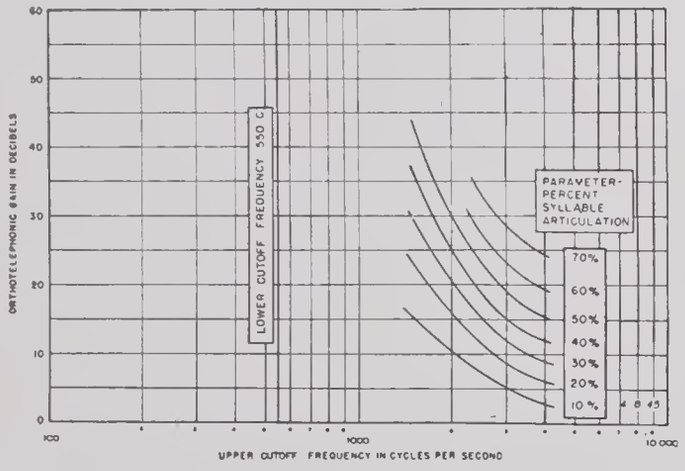


FIGURE 25. Equal-articulation contours for band-pass systems with fixed lower cutoff at 550 c. (Unfiltered spectrum B.)

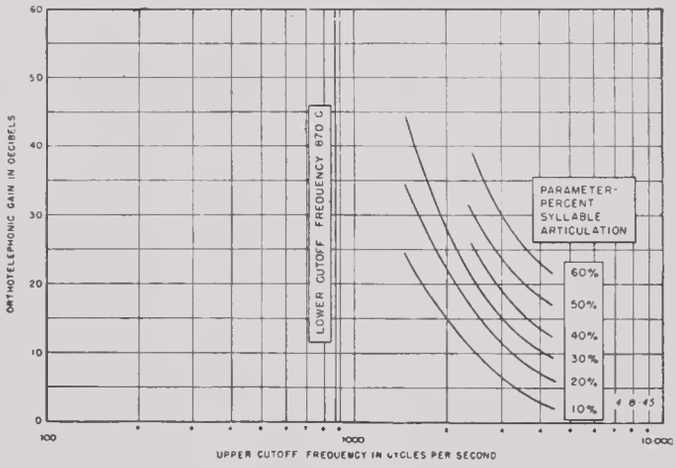


FIGURE 26. Equal-articulation contours for band-pass systems with fixed lower cutoff at 870 c. (Unfiltered spectrum B.)

The use of these contours may be illustrated by example. Suppose that a high-fidelity communication system is to be replaced by a system passing only the frequencies from 750 to 3,000 c. Suppose further that these systems are used

in a noise having a flat or falling spectrum similar to the types tested and that it is desired to determine the increase in gain required to maintain the articulation at 50 per cent. Then entering Figure 29 at the abscissas of 750 and 3,000 c shows that a gain of 15 db is required to maintain articulation at 50 per cent.[d] If, however, an increase of only 10 db is practicable, a band from 640 to 3,500 c is required to maintain 50 per cent articulation.

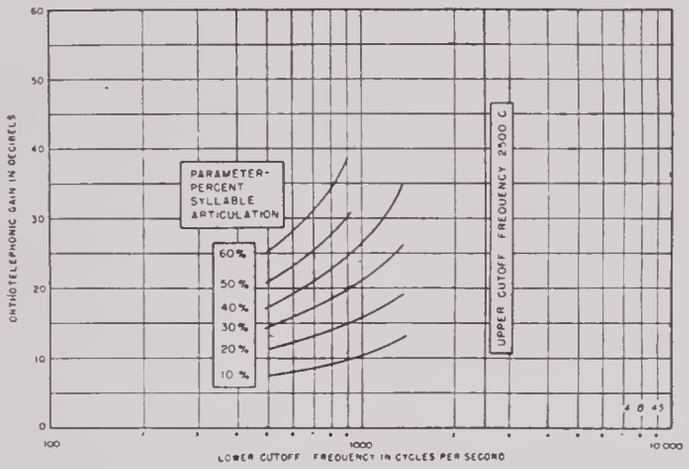


FIGURE 27. Equal-articulation contours for band-pass systems with fixed upper cutoff at 2,500 c. (Unfiltered spectrum B.)

It should be emphasized that these contours are intended to furnish only approximate results. They were obtained with idealized pass bands and should be generalized to other types of band-pass systems with some caution. They apply to those instances in which the spectrum of the masking noise extends fairly uniformly over the major part of the audio-frequency range. Since there is evidence that articulation is determined more by the speech-to-noise ratio than by the absolute level of the noise, the contours are not restricted to the particular noise and speech levels used in obtaining the data. By the use of proper transformations,[3] these results obtained with nonsense syllables will be valid for other types of test materials.

It may be stated as a general conclusion that an adequate speech-to-noise ratio should be provided over as wide a frequency range as possible. For ideal speech transmission the frequency range should extend from about 200 to 7,000 c, and the signal-to-noise ratio at each frequency should be 25 db or more.

Effects of Resonant Peaks

A special case of band-pass frequency distortion is provided by resonant earphones. The typical earphone resonance occurs between 1,000 and 2,000 c and can, therefore, be evaluated in terms of the generalized equal-articulation contours. The effects of resonant peaks above or below this frequency range have been studied in less detail, and results indicate that a low resonant frequency is more damaging to intelligibility than a high-frequency resonance.

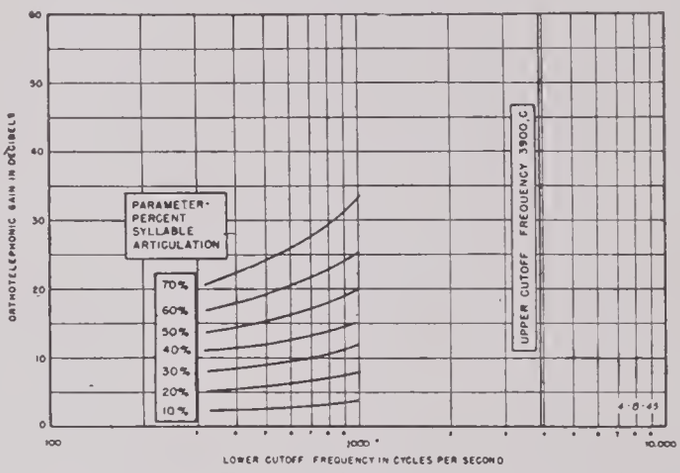


FIGURE 28. Equal-articulation contours for band-pass systems with fixed upper cutoff at 3,900 c. (Unfiltered spectrum B.)

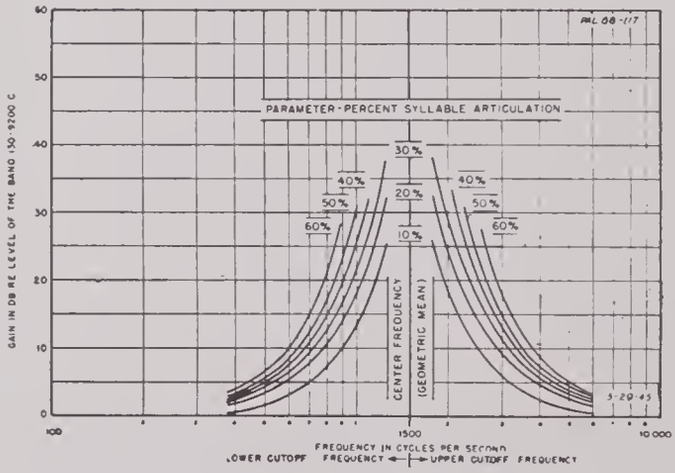


FIGURE 29. Generalized equal-articulation contours plotted for band-pass systems all having a center frequency of 1,500 c.

[d] If the center frequency is, for example, 1,200 c instead of 1,500 c, multiply the two cutoff frequencies by the ratio of 1,500 to 1,200 and enter Figure 29 with the resulting frequencies.

Illustrative data were obtained with the four response characteristics shown in Figure 30 (response measured with a 6-cu cm coupler). These characteristics provide a basis for comparing a relatively flat or nonresonant system with systems resonant at low (375 c), intermediate (1,050 c), and high (2,500 c) frequencies. Word-articulation results, using the check-list method with 90 db SPL of random noise, are shown in Figure 31. None of the resonant systems is as effective as the nonresonant system.

FIGURE 30. Single frequency-response characteristics for flat and resonant systems.

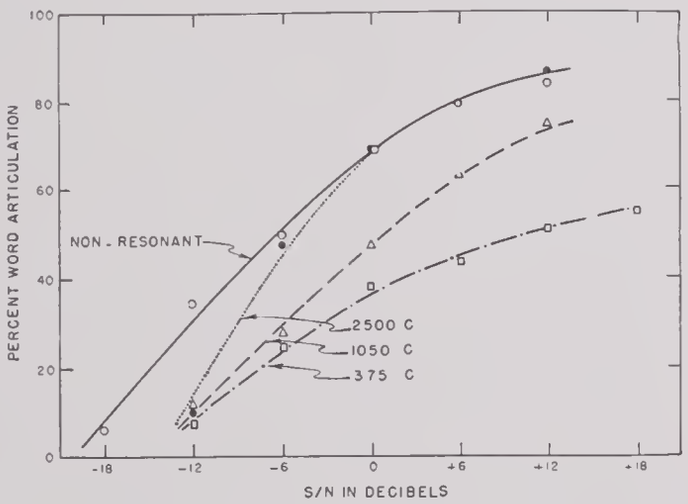


FIGURE 31. Articulation as a function of signal-to-noise ratio for resonant systems shown in Figure 30.

The most important objection to resonant systems, however, arises when communication is attempted in intense ambient noise. If the level of the speech through the system is high enough to override the noise, the level of the received speech at the resonant peak is painful to the listener. As a general rule, therefore, it is advisable to determine that the overall response of a communication system is free of prominent resonant peaks.

7.3 COMBINED FREQUENCY AND AMPLITUDE DISTORTION

In practice both frequency and amplitude distortion may occur in the same system, and it is necessary to know the importance of their interactions. The possible permutations of frequency distortions, amplitude distortions, and noise conditions are so numerous, however, that this line of investigation has really only begun. Consequently, this section can be only suggestive rather than definitive.

7.3.1 Effects of Tilting and Clipping

Frequency distortion of the type referred to in Section 7.2.3 as "tilting" was tested with various amounts of sharp, symmetrical peak clipping. The response characteristic of the entire system in the absence of nonlinear distortion is shown in Figure 19, and these can be conveniently referred to as HP-12 (high-pass system, rising 12 db per octave), HP-6, flat, LP-6 (low-pass system, falling 6 db per octave), and LP-12. The speech signal was first passed through the tilting network, then peak-clipped and fed to the listener's earphones. The clipping level was held constant at 112 db SPL, and no speech peaks above this level reached the listener's ears.

The results of word articulation tests made in quiet conditions over these systems are shown in Figure 32. Speech intensity is indicated in terms of the number of decibels the peak amplitude of the signal (as read on an oscilloscope) fell below the fixed clipping level. As the speech intensity is increased, the per cent word articulation rises to a maximum which is maintained up to the level at which clipping begins. Below the clipping point there is little difference between the tilts, although

LP-12 appears somewhat inferior. Above the clipping point, however, large differences appear in the susceptibility of the different tilts to peak clipping. The LP-12 and LP-6 tilts show a marked deterioration in intelligibility as the amount of peak clipping is increased. The flat and high-pass systems, on the other hand, remain intelligible with as much as 60-db clipping. Apparently a response characteristic which rises at about 6 db per octave gives the smallest loss in intelligibility with severe peak clipping.

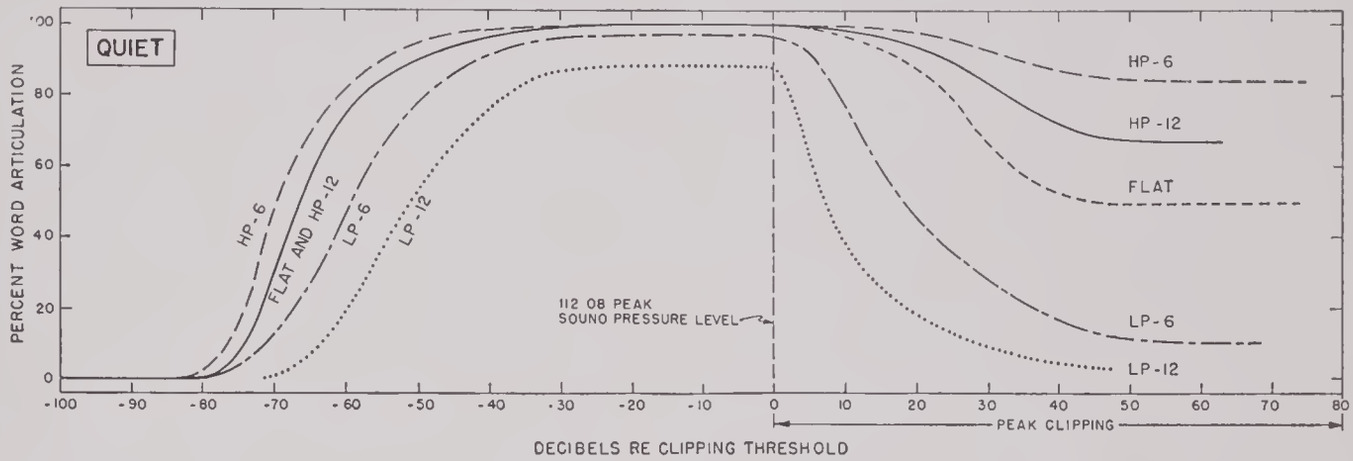


FIGURE 32. Word articulation in quiet as a function of speech intensity with frequency response as the parameter. Zero db (112 db SPL) represents the level at which clipping began.

Similar tests were run under noise conditions, using a dense, crackling static. The noise along with the speech was picked up by the microphone, and the talker monitored his speech level by a VU meter in such a way that a signal-to-noise ratio of +10 db was maintained at the microphone. Both speech and noise were then tilted and clipped; the amount of peak clipping was measured from the point at which the combined signal was limited. The results of these tests are shown in Figure 33 for the HP-6, flat, and LP-6 tilts. In the absence of nonlinear distortion the flat response is superior, but when clipping is introduced the HP-6 tilt gives slightly higher articulation scores.

The choice of 112 db as the clipping level in these tests is arbitrary and does not greatly affect the experimental results. Essentially similar functions were obtained with higher clipping levels.

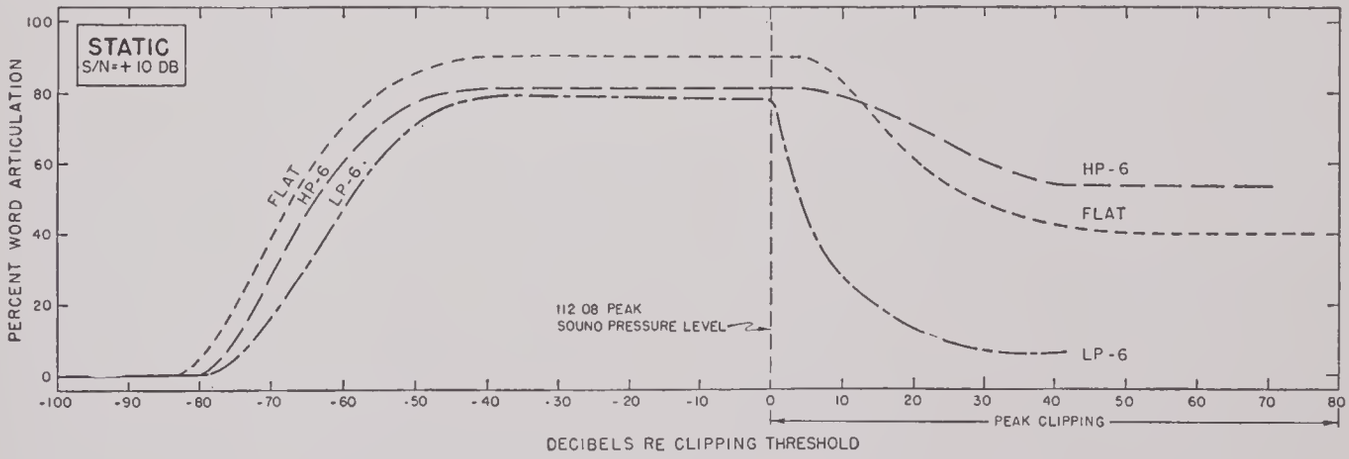


FIGURE 33. Word articulation in noise as a function of speech intensity with frequency response as the parameter. The signal-to-noise ratio at the microphone was +10 db.

7.3.2 Application to Speech at High Altitudes

Two special problems arise when communication is attempted at high altitudes (see Chapter

9 and Section 4.1.3). The overall speech level drops considerably, and the quality of the speech deteriorates. Changes in the overall level as a function of altitude necessitate either some sort of altitude-controlled amplification or a compression amplifier to reduce the range of differences. Also quality changes due to reduced ambient pressure and to the oxygen-mask enclosure suggest that some low-frequency attenuation might be desirable, especially when carbon microphones are used. In order to explore these possibilities, tests were run using a variety of conditions of frequency and amplitude distortion.

Several voices were phonographically recorded for test. Talkers with A-14 oxygen masks were located in an altitude chamber, and speech samples were obtained at 5,000 ft (sea level) and at 35,000 ft, using a carbon microphone (ANB-M-C1). Frequency distortion was introduced in the playback amplifiers. The 28 different response characteristics tested are

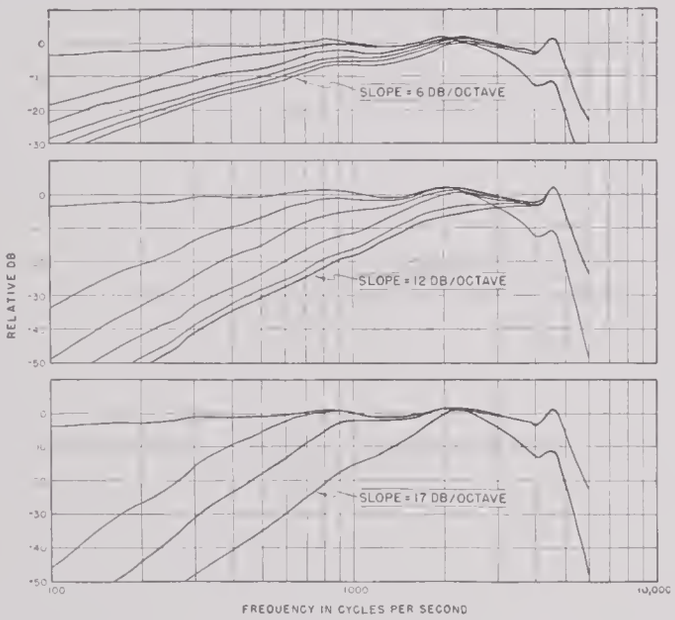


FIGURE 34. Single frequency-response characteristics used in testing the intelligibility of speech at high altitudes.

indicated in Figure 34. The low-frequency cutoff varied from 500 to 4,000 c, and the slope of the cutoff was 6, 12, or 17 db per octave. The two high-frequency cutoffs were 3,200 and 5,000 c. Random noise interference was introduced electrically after the speech signal had been filtered, and the intensity of the noise was held constant at a level corresponding to 110 db re 0.0002 dyne/cm^2. The peak speech intensity was determined with a calibrated cathode-ray oscilloscope. The signal-to-noise ratio is expressed as the difference in decibels between the peaks of the speech and the rms level of the noise.

Results obtained in the absence of peak

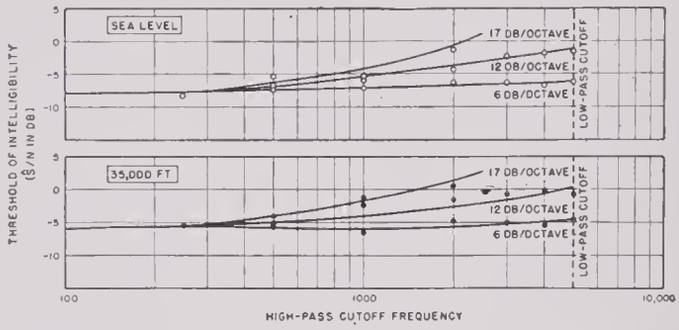


FIGURE 35. Showing threshold of intelligibility as a function of the high-pass cutoff frequency, with slope of cutoff as parameter. Low-pass cutoff was 5,000 c.

clipping are shown in Figure 35 where the signal-to-noise ratio at the threshold of intelligibility is plotted in decibels against the low-frequency cutoff with the slope of the cutoff as the parameter. The two high-frequency cutoffs tested gave similar results.

Accentuation of the high-frequency components of the speech does not lower the signal-to-noise ratio at the threshold of intelligibility (see Section 7.2.3). The quality seems better, however, as the "booming" caused by the mask enclosure is reduced. As can be seen in Figure 35, the effect of the high-pass filter is most pronounced for the steeper cutoff slopes.

However, little or no improvement in the intelligibility of altitude speech is obtained by deliberate frequency distortion. It is probable that the altitude, the mask enclosure, and the extraneous valve noises have more extensive effects upon speech sounds than is indicated by frequency-response characteristics. Consequently, simple equalization of the response characteristic does not produce a dramatic improvement in performance.

Tests were next conducted with the addition of peak clipping to the frequency distortion. The altitude speech was first filtered to produce the desired frequency distortion, and then the

speech peaks were clipped 0, 10, 20, or 30 db. Random noise was mixed electrically with the peak-clipped speech, the intensity of the noise being constant at a level corresponding to 90 db re 0.0002 dyne/cm^2. The check-list method of articulation testing was used.

In Figure 36 the articulation obtained with frequencies. If this signal is now peak clipped, the intelligibility of the speech declines in a manner similar to the decline shown in Figure 32 for the low-pass conditions. When the low frequencies are attenuated by a 500-c high-pass filter, however, peak clipping does not significantly change the intelligibility of the

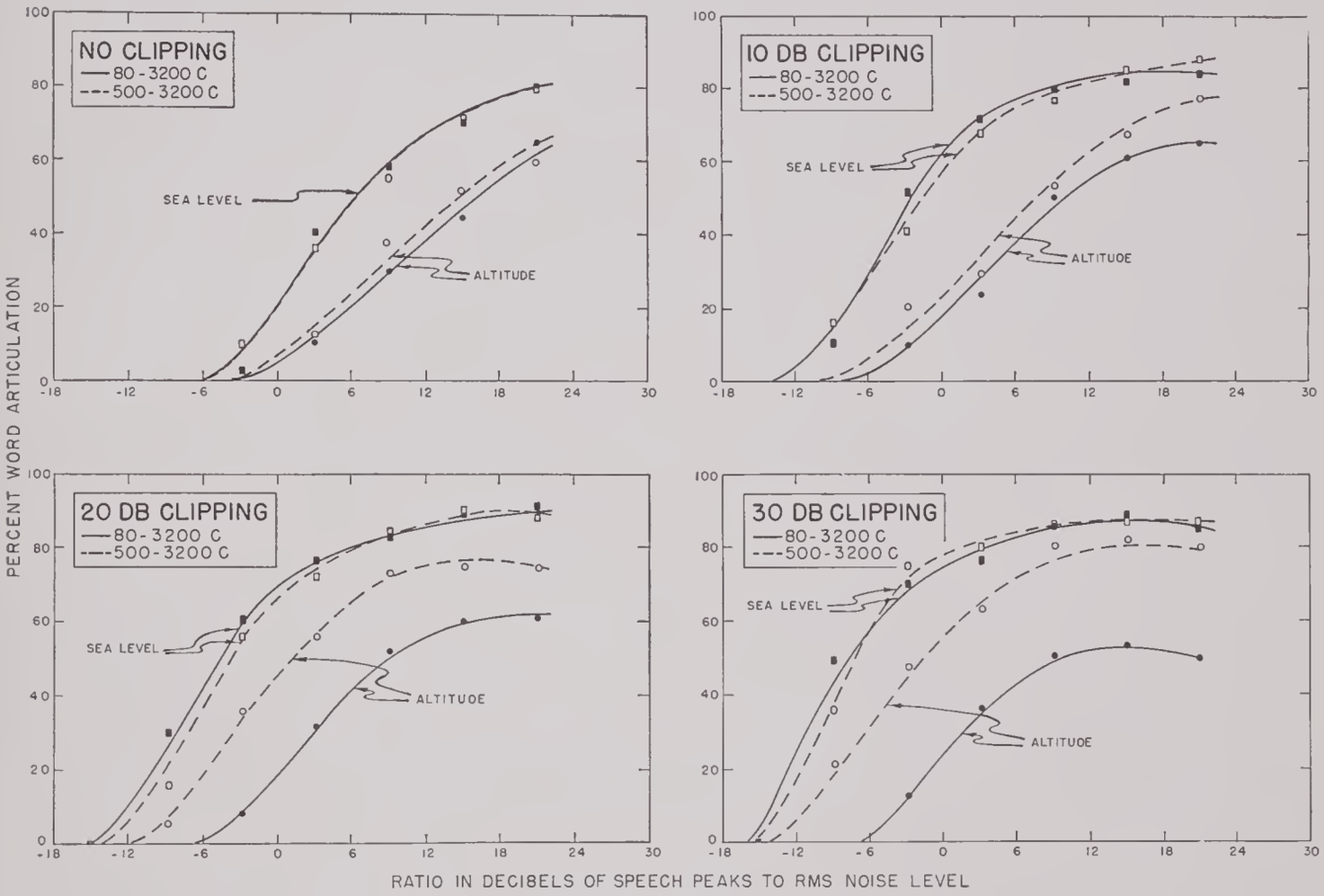


FIGURE 36. Word articulation as a function of peak-speech-to-rms-noise ratio with altitude, frequency response, and peak clipping as parameters. A carbon microphone (ANB-M-C1) was used in the oxygen mask (A-14).

a low-frequency cutoff at 80 c is compared with results obtained using a 500-c cutoff. In the absence of peak clipping, the attenuation of frequencies below 500 c does not seriously affect articulation scores either at altitude or sea level (see Figure 35). This is also true for sea-level speech when clipping is introduced. When the high-altitude speech is peak clipped, however, the effect depends in large part upon the frequency response of the system. When a flat system is used, the mask and carbon microphone combine to accentuate the low altitude speech relative to the intelligibility of sea-level speech.

To apply these data to the practical problems of communication in and between airplanes, account must be taken of the fact that the signal intensity at altitude is about 10 db below the intensity at sea level. If a clipping amplifier were designed, for example, to clip sea-level speech 30 db, the speech at altitude would be clipped only about 20 db. A comparison of the appropriate functions in Figure 36 (20-db clipping at altitude vs 30-db clipping at

sea level) shows that about 8 db more amplification is needed at altitude to reach 50 per cent articulation $\hat{S}/N = +1$ db) than is needed at sea level ($\hat{S}/N = -7$ db). This compares with 15 to 20 db of additional amplification required at altitude to give equivalent articulation scores when peak clipping is not used. Thus the range of signal levels which the interphone or modulator is required to handle is substantially reduced.

Peak clipping has a further advantage in the case of AM radio-telephony at high altitudes, because the clipping level can be adjusted to give 100 per cent modulation of the carrier at all times. The disadvantage arises from the very severe clipping of sea-level speech which is required in order to obtain adequate limiting of the weaker speech signal at altitude. Compromises are obviously necessary.

Results reported in this chapter have indicated that it is often possible to take liberties with speech as the talker produces it, and that distortion, particularly nonlinear distortion, is not always undesirable. The question which arises for the concern of future investigators is, therefore, what changes can be made in a typical speech signal to put it into optimal form for any given communication situation. Can speech be packaged to fit more neatly into available communication equipments? The problem of finding the best speech package for every situation is an extensive undertaking.

Chapter 8

INTELLIGIBILITY OF SPEECH: TYPES OF INTERFERENCE

MILITARY COMMUNICATIONS are normally subject to a wide variety of interfering noises. A ship-to-plane radio-telephone link, for example, may be simultaneously afflicted by pulse interference from shipborne radar and other electrical equipments, atmospheric static, tube noise in the radio receiver, and the roar of the plane's engines or the ship's guns. It is reasonable to suppose that some of these noises may be tolerable while others provide a serious cause for concern. It is necessary, therefore, to explore an extensive range of signals to determine what factors contribute to their effectiveness for masking speech.

8.1 GENERAL CONSIDERATIONS

As far as the ear is concerned, the important dimensions of an interfering noise are its intensity, spectrum, temporal continuity, and annoyance value. The most important of these is intensity, for if the noise is 20 or 30 db less intense than the speech signal, other aspects of the interference become irrelevant. With a given interference of sufficient intensity, however, the amount of masking depends upon the spectrum and the temporal continuity of the noise.

Masking is customarily expressed in terms of the shift in the threshold of hearing which results when an interfering signal is introduced (see Section 3.4). However, the signal-to-noise ratio can also be used as a measure of the relative effectiveness of different types of interference. In this sense the masking efficiency of a signal is defined in terms of the percentage of the words which cannot be heard at a given signal-to-noise ratio. Thus it is possible to state, for example, that noise A masks the same percentage of words at a signal-to-noise ratio of 2 db as noise B masks at a signal-to-noise ratio of 10 db, i.e., noise A is 8 db more effective than noise B in masking speech.

8.1.1 Spectrum of the Interfering Noise

The effective component frequencies of speech are those ranging between 200 and 7,000 c and, in general, a signal whose spectrum is correspondingly wide has the maximum masking efficiency. Before considering these complex sounds, however, it is useful to review some of the information available on the masking of pure tones by pure tones.

The simplest case of masking is encountered when two pure tones are introduced into the ear and the observer is instructed to adjust the intensity of one of these tones until it is just detectable.[1] It is found that the presence of a second tone tends to reduce the ability of the ear to hear the first tone and it is, therefore, necessary for the intensity of the first tone to be increased before the listener can detect it. How much the intensity must be increased will depend upon the intensity of the second or masking tone and also upon the frequency relation between the two tones. The experimental results lead to two general conclusions. (1) The masking produced by a tone is greater for frequencies above the frequency of the masking tone. (2) The spread of the masking to the higher frequencies is most pronounced at high intensities of the masking tone. These effects are due, at least in part, to the nonlinear operation of the ear at high intensities.

In general, the masking effectiveness of a narrow band of noise is also greatest for frequencies in or above the band, and the masking increases rapidly at high intensities. When the masking signal is a band of noise, however, both the center frequency and the width of the frequency band determine the masking effectiveness. If the masking signal covers a narrower range of frequencies than the masked signal, the frequencies of the masked signal which lie outside the range of the masking signal may be heard. If the spectrum of the masking signal is much wider than that of the signal to be masked, the outlying

frequency components of the masking signal do not contribute to the masking. Logically, there must be some optimal spectrum for the most efficient masking of a given signal, and this spectrum will depend upon the range of frequencies in the masked signal.

The problem is, therefore, to apply these general principles derived from pure-tone masking to the masking of the speech spectrum. If pure tones are used to mask speech, it is obvious that tones of low frequency will be more effective than tones of high frequency, because at high intensities the masking effects of a low-frequency tone will extend upward in frequency to cover the entire range of speech. From the general principle that the optimum frequency range of the masking signal is a function of the frequency range of the signal to be masked, it is apparent that even the most effective single frequency will not efficiently mask the wide frequency range of speech. If, therefore, bands of noise are used to mask the speech, the masking efficiency will be improved. The frequency relations between the masking and the speech signals are such that the maximum efficiency is produced by masking signals which cover the frequency range of speech and which have the maximum interfering intensity concentrated at the low end of this range.

It should be noted that, within certain broad limits, the masking efficiency of a noise is independent of the phase relations among the various component frequencies making up the spectrum. The actual form of the masking wave as seen on an oscilloscope is of little consequence in determining the masking efficiency. This agrees with the general notion that the ear is relatively insensitive to phase relations.

8.1.2 Temporal Continuity of the Interfering Noise

The nature of speech and the operation of the ear are such that relatively large portions of the speech can be completely "blanked out" in time without seriously lowering the intelligibility of the speech.

This generalization may be experimentally demonstrated in either of two ways. The speech signal can be turned on and off intermittently, and the loss in intelligibility can then be expressed as a function of the percentage of time the signal was available to the ear. Or an interfering noise may be intermittently interrupted and the loss in intelligibility can be expressed as a function of the percentage of the total time during which the speech signal is not masked by the noise.

When an electronic switch was used to interrupt speech at a rate of nine interruptions per second, little loss in word articulation was obtained until relatively large portions of the speech were missing. When only 25 per cent of the speech pattern was present it was still possible to understand approximately half the words. This relation is shown in Figure 1.

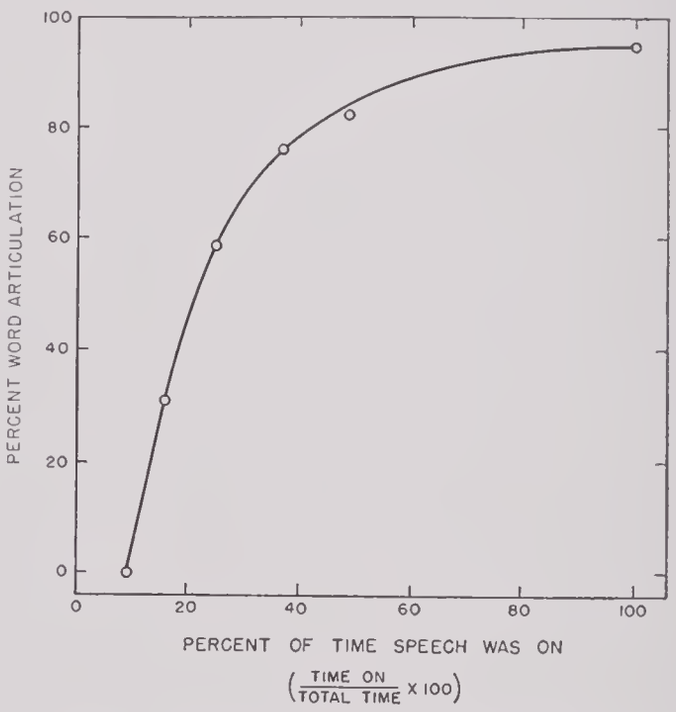


FIGURE 1. Showing increase in intelligibility as a function of the per cent of the time the speech wave is unattenuated. The switching rate was nine interruptions per second.

In the case where the speech was continuous but an interfering noise was periodically interrupted at a rate of nine times per second, the masking efficiency was found to be directly related to the portion of the time the noise was on. The relation between articulation and the intensity of the noise is shown in Figure 2, where the percentage of the time the noise

is on is taken as the parameter. A noise which is present only 25 per cent of the time produces practically no masking of speech.

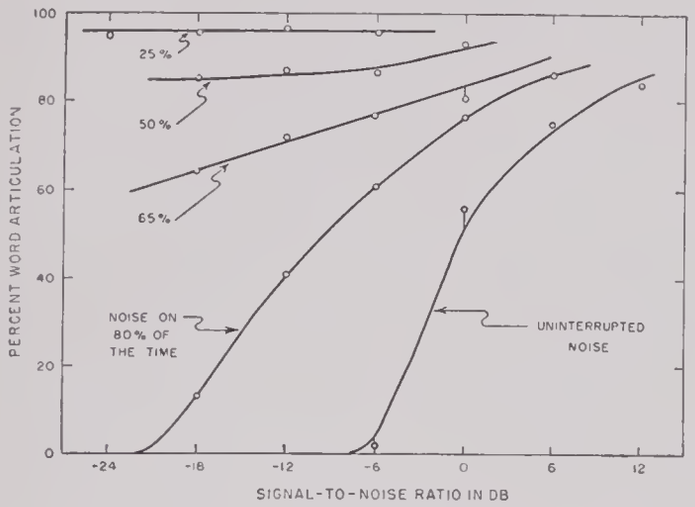


FIGURE 2. Showing the masking efficiency of a noise signal which is interrupted for different proportions of the time at a rate of nine interruptions per second.

8.1.3 Annoyance Value of the Interfering Noise

It is obvious, of course, that the masking efficiency of a signal is more basic to its interference with communications than is its annoyance value. Annoyance might force a listener to take off his headphones, but the probability of such an event cannot be predicted from laboratory tests. It should be noted, however, that many signals, although of little masking effectiveness, do have characteristics which are unpleasant and annoying. In general the annoyance value of a signal increases as the loudness is increased, the pitch is raised, or the signal is changed irregularly in pitch or loudness (see Section 2.4). Signals having such characteristics will usually not prevent a determined listener from receiving the message, but from the point of view of listener efficiency such noises should be avoided wherever possible.

8.2 TONAL INTERFERENCE

To classify the various types of signals which have been investigated, at least three categories of sounds are useful: tones, noises, and voices. This rather arbitrary classification of the various sounds serves to organize the following discussion of the different types of interfering signals.

8.2.1 Pure Tones

Simple sinusoidal tones do not effectively mask speech. Since such interference is sometimes encountered, however, it is worth while to determine what frequencies have the most deleterious effects upon intelligibility. The relation between the frequency of the tone and the shift in the threshold of perceptibility, with the intensity of the tone as a parameter, is demonstrated in Figure 3. For these determinations, dynamic earphones (Permoflux PDR-10) in supra-aural cushions (MX-41/AR) were used, and the intensity of the tone was measured by the voltage applied to the earphones. The bottom curve of this family (— 50 db re 1 v peak to peak across the earphones) corresponds approximately to a sound-pressure level of 62 db re 0.0002 dyne/cm^2. At this level a pure tone has a negligible effect upon the threshold of intelligibility for continuous speech. As the intensity of the tones is increased above this level, however, the shift in the threshold of perceptibility becomes considerable. The occurrence of masking only at high intensities is understandable in terms of the overloading of the ear and the resultant nonlinear distortion. With the introduction of overtones, the effective frequency coverage of the masking signal is increased.

It will also be noted that the maximum masking effectiveness is obtained at different frequencies, depending upon the intensity of the tones. At the lowest intensities the maximum effectiveness is obtained with tones in the neighborhood of 500 to 600 c. For the more intense tones, however, the maximum effectiveness is obtained with frequencies between 300 and 400 c.

The earphone cushions permitted a small leak which lowered the low-frequency response of the earphones when coupled to the listener's ear. The dotted curves have been added to indicate the masking which would presumably be obtained if a constant sound-pressure level

were generated. These corrections are based upon threshold calibrations and can be rather accurately made for sine waves. In the following discussion of complex tones, however, the effect of cushion leaks is not taken into account.

3.2.2 Complex Tones

When speech is masked by complex tones, the role of the aural harmonics introduced at high levels by distortion in the ear is less sig-

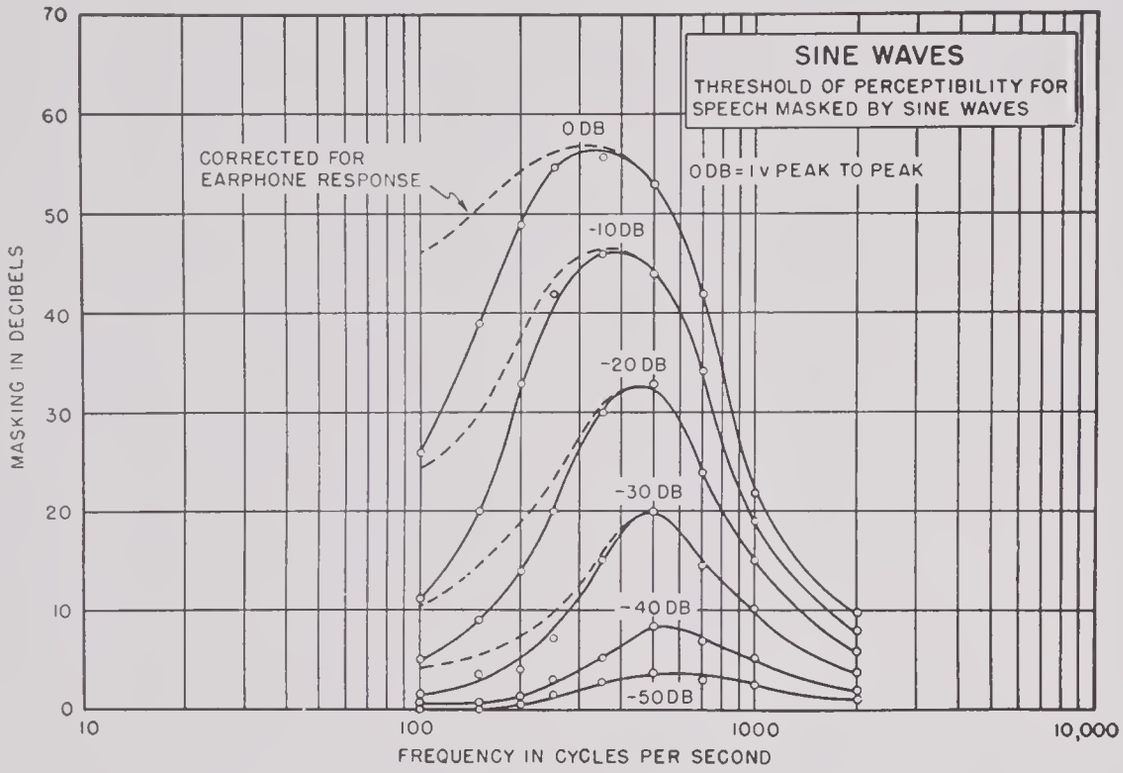


FIGURE 3. Showing the masking of speech by sine waves of various frequencies, with the level of the sine wave as the parameter.

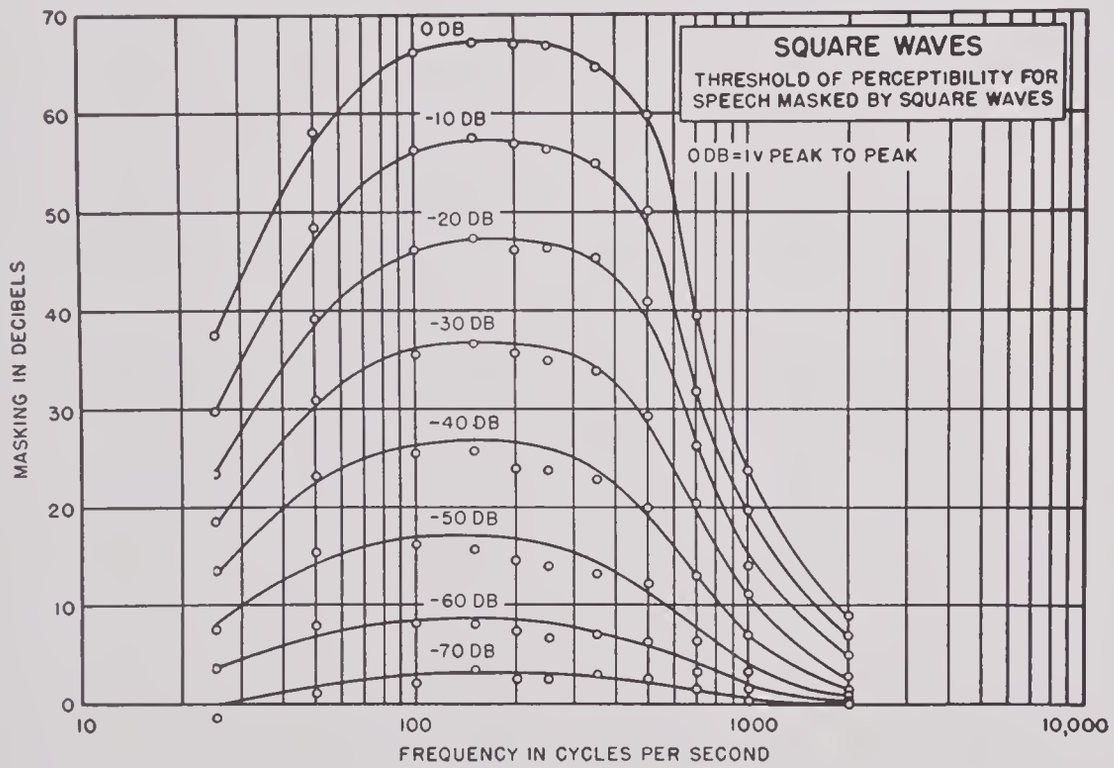


FIGURE 4. Showing the masking of speech by square waves of various frequencies, with the level of the square wave as the parameter.

nificant. With square waves or trains of pulses the harmonic content of the signal is sufficiently great that the introduction of further harmonics by the ear contributes relatively little to masking.

The shift in the threshold of perceptibility of speech masked by square waves of different frequencies is shown in Figure 4, with the intensity of the square wave as the parameter. As in the case of the pure-tone masking signal, the high frequencies are relatively ineffective masking agents. At low frequencies, greater masking is produced by a square wave than by a pure tone. A square wave with the sound-pressure level corresponding to approximately 115 db, measured in a 6-cu cm coupler, shifts the threshold of perceptibility approximately 68 db.

Similar results are obtained if trains of short pulses are used. Figure 5 shows the results of tests using a pulse 10 μsec long. For such a train of pulses all the multiples of the fundamental frequency are present in amplitudes equal to that of the fundamental, and, consequently, the pulse signal covers effectively the entire range of speech frequencies. Lowering the frequency of the train of pulses produces greater separation in time between successive pulses but simultaneously produces less separation between the component overtones. At very low frequencies, therefore, the frequency coverage is excellent but intermittent, and, when the successive pulses are separated by too long an interval, speech can be heard between the successive pulses, and the masking efficiency falls rapidly.

Figures 3, 4, and 5 are directly comparable in terms of the peak voltages applied to the earphones. In all three cases, the top curve (0 db) is obtained with a signal of 1 v peak to peak across the dynamic transducers (Permoflux PDR-10). For the sine and square waves the sound-pressure levels quoted are the equivalent levels generated in a 6-cu cm coupler. For pulses of constant amplitude, however, the rms pressure varies directly with the pulse repetition rate. When the peak is held constant, the rms pressure rises 3 db per octave as the pulse repetition frequency [PRF] is raised. Consequently, this correction must be made if the curves are to be compared on the basis of equivalent sound-pressure levels. So compared,

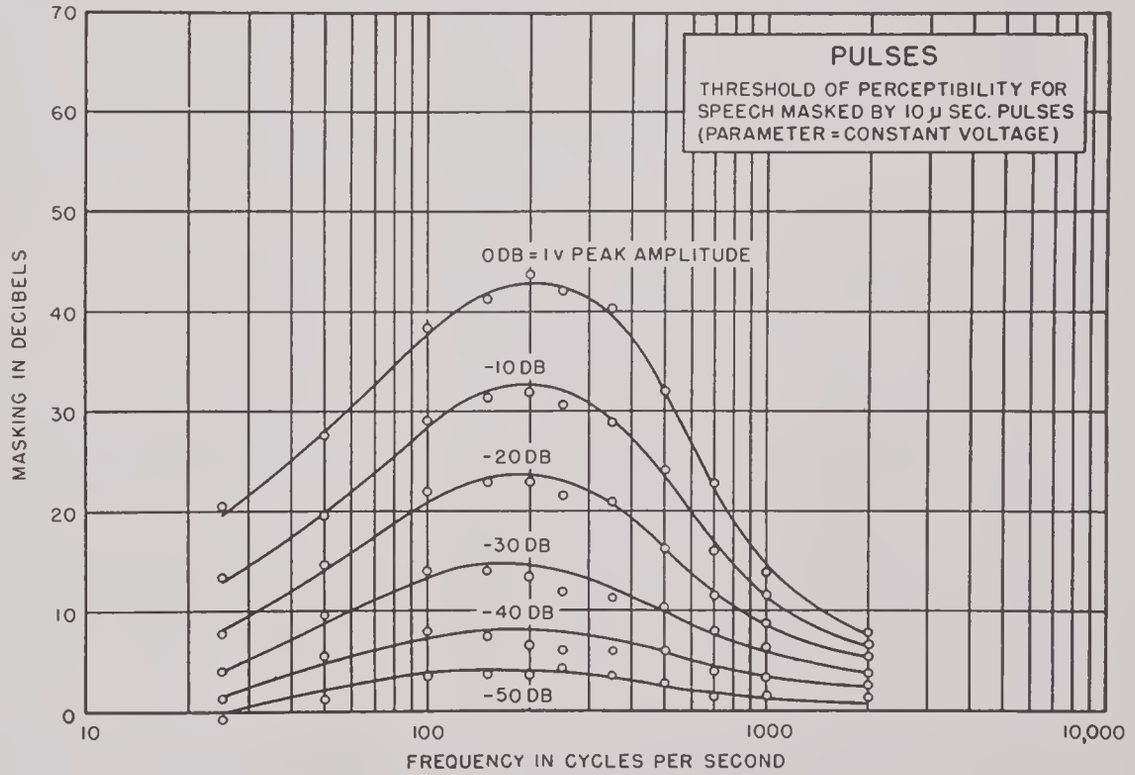


FIGURE 5. Showing the masking of speech by 10-μsec pulses at various pulse repetition frequencies, with the peak level of the pulse as the parameter.

however, the pulses have the greatest and sine waves the least masking effect.

8.2.3 Patterns of Tones

In addition to pure or complex tones of continuous character, signals are sometimes encountered which provide a pattern of tones of different frequencies. In general there are three types of patterns which may be encountered: stepped tones, warble tones, and musical signals.

Stepped-Tone Signals

At least eight variables can be distinguished for the type of tonal signal which switches abruptly from one frequency to another. Each of these variables was experimentally investigated[3] to determine (1) its contribution to the annoyance produced by stepped signals, and (2) its role in determining the masking efficiency of the signal. The eight variables, along with their effects on the annoyance and masking effectiveness of the signal, are summarized in Table 1.

Warble-Tone Signals

A similar study was made of the various aspects of a tonal signal which changes gradually—warbles—from one frequency to another. The results for this type of tonal pattern are summarized in Table 2.

Musical Signals

Recordings of popular dance music were also studied for their masking effectiveness. The masking of speech is not great, because the speech can be heard in the pauses between musical phrases and because clarinet and trumpet solos usually fall above the frequency range which produces the most effective masking. When tests were run with two selections played simultaneously through a compression amplifier (about 10 db of compression reduces the range of variation in the signal level), the masking effectiveness of the music was approximately equivalent to the masking produced by the stepped and warbled signals.

The results with these tonal patterns are consistent with the general considerations discussed in Section 8.1. Speech is masked most

Table 1. Summary of results for stepped-tone interference.

Variable studied	Effect on annoyance value	Effect on masking efficiency
Frequencies of the stepped tones	High frequencies are more annoying than low frequencies.	Low frequencies produce more masking of speech than do high frequencies.
Frequency range between highest and lowest stepped tones	Wide range of frequencies tends to increase the annoyance value.	A wide range of frequencies must involve some high frequencies, thus lowering the masking efficiency.
Addition of other steady tones	Decrease in annoyance, unless the steady tone is near enough in frequency to stepped tones to produce dissonance and beats.	Increases frequency coverage, gives better masking, especially if it is of low pitch and rich in overtones.
Complexity of wave form	Sharp wave forms are more unpleasant than sinusoidal waves.	Complex waves have more overtones, cover wider frequency range, increase masking efficiency.
Number of stepped tones in pattern	Little effect, although many steps are slightly less annoying than only three or four steps.	No effect, as long as all steps individually are of equal masking efficiency.
Number of repetitions of pattern per minute	Slow rates are slightly more annoying than rapid rates.	No effect.
Relative durations of each stepped tone	Rhythms may become more distracting than equal steps. Random changes in the switching speed increase the annoyance.	No effect.
Length of switching interval elapsing between stepped tones	A switching click is annoying. The presence of a silent interval in the pattern increases the annoyance.	For optimal masking, no silent interval should occur between steps. Masking is increased by reducing the switching interval to a minimum.

TABLE 2. Summary of results for warble-tone interference.

Variable studied	Effect on annoyance value	Effect on masking efficiency
Center frequency	High frequencies are more annoying than low.	Low frequencies mask speech more effectively than high.
Frequency range	A wide range is slightly less pleasant than a narrow range.	A wide range is apt to include some inefficient frequencies and thus may lower the masking effectiveness.
Rate of warble	Slow rate is slightly more annoying.	Slow rate is slightly more effective.
Manner of change	Irregular or random warbles are most annoying.	No effect as long as frequency range is relatively narrow.
Addition of steady tone	Steady tones tend to lower the annoyance value.	Tones added above or below the warble frequencies improve the frequency coverage and increase masking effectiveness.

effectively when the fundamental frequency of the masking signal is in the range from 200 to 500 c and when the signal is continuous in time and rich in harmonics. Under these conditions, tonal signals are only 2 or 3 db less effective for masking than random ("white") noise and are much more annoying to the listener.

8.3 NOISE INTERFERENCE

We have seen that the maximum interference with speech communication is produced by a signal which continuously blankets the frequency range of speech. The signal which is most likely to provide this continuous coverage and which is a standard hazard in military communication is noise, i.e., interference devoid of tonal character.

8.3.1 Modulated Pulses

A comparison between tonal and noise interferences can be obtained by the use of modulation. If we start with a "tonal" signal consisting of a steady train of pulses, it is possible to

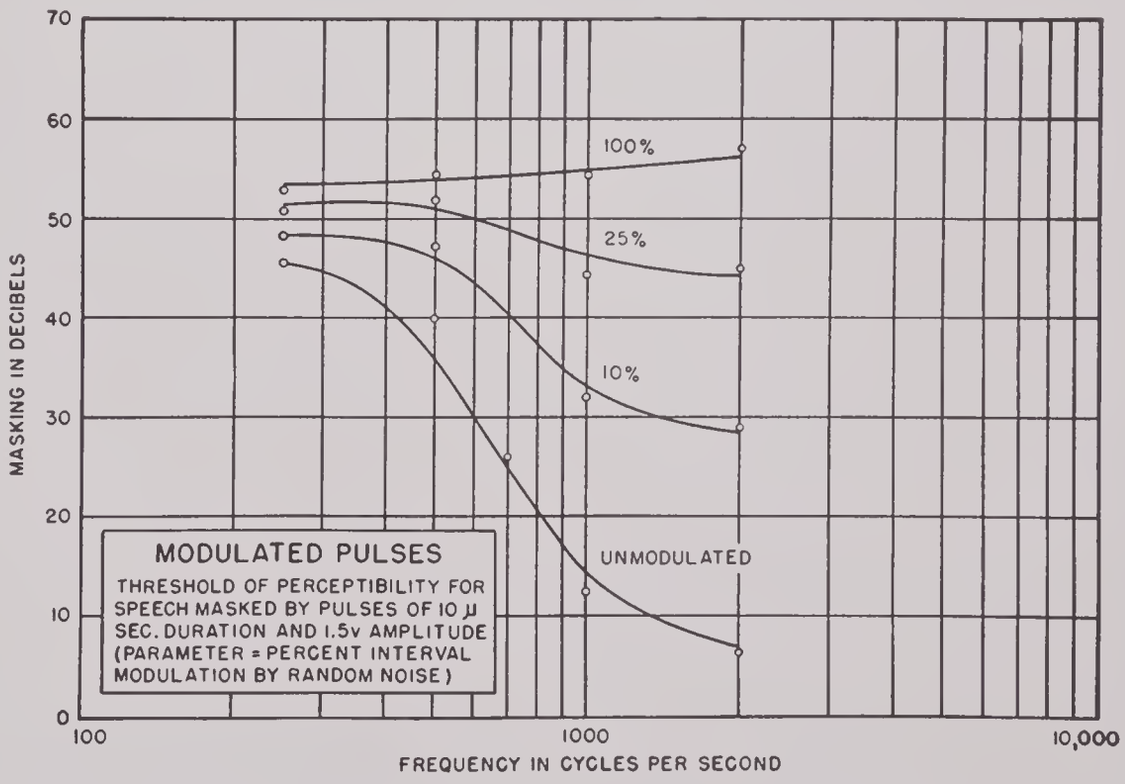


FIGURE 6. Showing the masking of speech by modulated pulses of 10 μsec duration, with per cent interval modulation as the parameter.

change the intervals between pulses by modulation with random noise, and to increase the modulation until the steady buzzing quality of the pulses comes to resemble the noisiness of very dense static (see Section 3.4.3). The qualitative change in the pulses is accompanied by a simultaneous increase in the masking effectiveness of the signal. The amount of increase in masking depends upon the frequency of the steady train of pulses, the increase being greatest for high frequencies.

Results of threshold articulation tests run under these conditions are shown in Figure 6. The peak voltage of 10-μsec pulses was held constant at 1.5 v, and the amount of modulation was varied for frequencies between 250 and 2,000 c. The amount of modulation was expressed in per cent of the modulated frequency. Thus, a tone of 1,000 pulses per second [pps] is said to be modulated 10 per cent if the shortest and longest intervals between pulses correspond to PRF's of 900 and 1,100 pps. The noise used to modulate the pulses consisted of a band of frequencies from 20 c to half the pulse repetition rate.

Inspection of Figure 6 shows that a very slight amount of modulation greatly increases the masking. The increased masking is presumably attributable to the more uniform coverage of the frequency range by the modulated signal. A little interference at all frequencies is apparently more damaging to speech communication than a great deal of interference concentrated at only a few frequencies.

8.3.2 Random Noise

A more representative type of noise encountered in practice is random, or "white" noise. Considerable experimental evidence has accumulated concerning the effects of both ambient and electrical random noise, and the attempt has been made to determine the minimum speech-to-noise ratio which will allow accurate perception of speech under conditions similar to those encountered in military aviation. This minimum ratio depends upon the character of the noise, the level of the speech, and the amount of distortion in the equipment. Over a wide range of conditions, however, a speech-to-noise ratio of + 6 db permits the listener to receive connected discourse with reasonable accuracy. However, when listening to isolated words at this signal-to-noise ratio, the listener correctly perceives about ten words per hundred fewer than with a more favorable ratio. For this reason the specification of a tolerable level of background noise should probably be set no less than 6 db below the level of the received speech.[2]

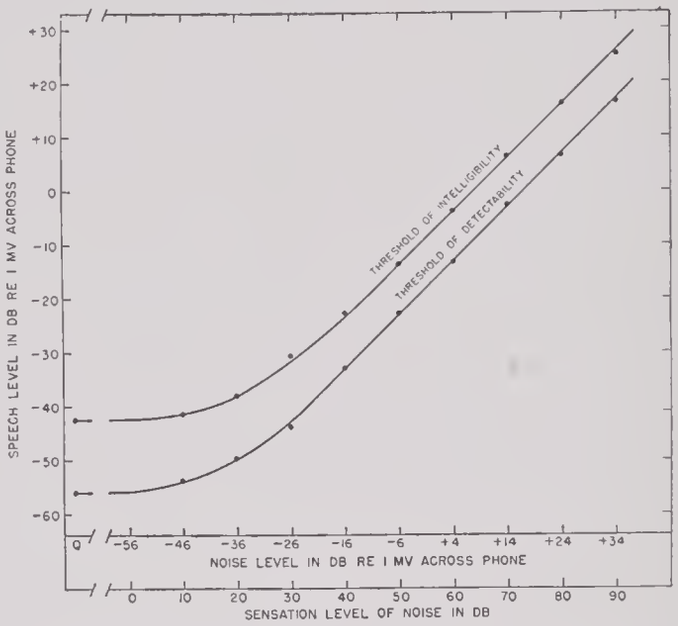


FIGURE 7. Showing the effect of the overall level of the white noise on the thresholds of intelligibility and of detectability of speech.

TABLE 3. Comparison of the masking of speech and of tones by random noise.

Sensation level of noise	Average shift in threshold of detectability and threshold of intelligibility	Average shift in threshold for tones, 500, 1,000, and 2,000 c
10	1.4	...
20	5.3	3.1
30	11.8	12.4
40	21.0	21.3
50	30.7	31.2
60	40.5	40.7
70	50.9	51.0
80	60.4	60.8
90	69.8	69.7

Threshold articulation tests run with a linear, wide-band system using random noise to mask speech[4] typically give the results shown in

Figure 7. These data represent the thresholds of intelligibility and of detectability for different sensation levels of the noise. Above a sensation level of 30 db of noise (corresponding to sound-pressure level of approximately 0 db per cycle), a 10-db increase in the noise results in a corresponding 10-db rise in the masking.

Upon comparison of the masking of speech by noise (see Figure 7) and the masking of tones by noise (see Figure 10 in Chapter 3), it is obvious that the masking effects are similar. This correspondence is particularly close if the average masking at 500, 1,000, and 2,000 c is compared with the average shift of the thresholds of intelligibility and of detectability. Table 3 shows the correspondence.

8.3.3 Narrow Bands of Noise

By the use of highly discriminative filters it was possible to obtain narrow bands of noise. each band. The results of these tests are shown in Figure 8, along with the frequency limits of the bands of noise used. No single band of frequencies tested was more effective in masking speech than was unfiltered noise. Bands of noise above 1,000 c failed to mask completely all the speech sounds.

These results for narrow bands of noise agree in general with the results obtained with tonal signals used to mask speech, for in both cases sounds of lower frequency (from 100 to 1,000 c) are more destructive to intelligibility than are the higher frequencies. Considerations of this sort led to the prediction that the noise which would produce the greatest interference with speech communications would have energy at frequencies between 100 and 4,000 c, but with most of the energy concentrated below 1,000 c. Actual experimentation with a variety of wide-band noise spectra confirmed this ex-

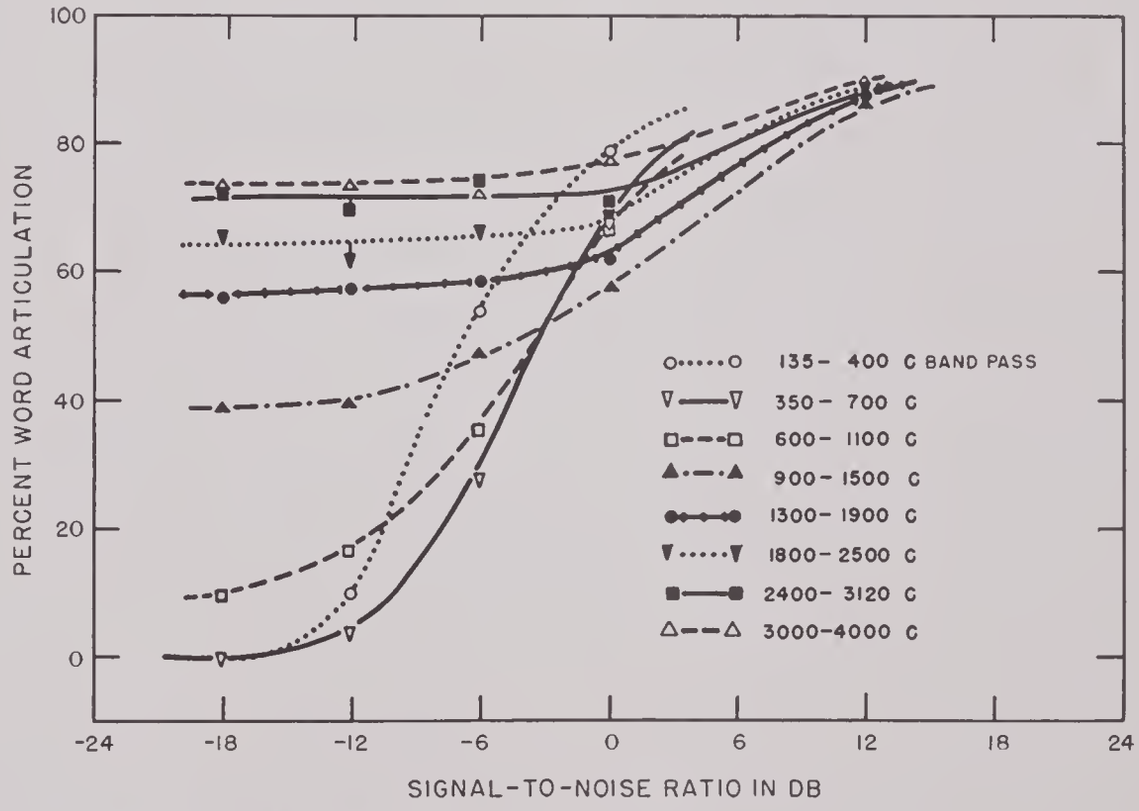


FIGURE 8. Relating per cent word articulation to noise level with the different noise bands as the parameter. The level of the speech was 95 db re 0.0002 dyne/cm^2.

Eight bands, continuous in frequency from 135 to 4,000 c, were used to mask speech, and articulation scores were obtained by the check-list method for several signal-to-noise ratios for pectation[3a] and placed the maximum energy per cycle at approximately 500 c. The level per cycle can drop as much as 15 db between 1,000 and 4,000 c. A comparison of this noise spec-

trum with the long-time average spectrum for speech (see Figures 1 and 2 in Chapter 4) reveals a marked resemblance. The tentative generalization can be made, therefore, that the maximum interference with speech is produced by a noise spectrum which provides a constant speech-to-noise ratio at all frequencies.

8.4 VOCAL INTERFERENCE

The masking noise most apt to produce a constant speech-to-noise ratio at all frequencies is, of course, speech itself. Fortunately, however, the analytical ability of the ear and the intermittent character of speech make it relatively simple to distinguish one voice from another. Serious interference results only when the desired voice is masked by a babble of other voices.

Articulation tests using 1, 2, 4, 6, and 8 interfering voices showed that masking is greatest when 4 to 8 voices are used. A single voice is 10 to 12 db less effective than a complex babble. The maximum masking obtained with vocal signals is equivalent to the masking obtained with the noise spectrum (described above) which produces the greatest loss in intelligibility.

In general, the meaning of the interfering speech, and even the language, are not important factors in determining the masking effect. On the other hand, laughter and boisterous shouting are somewhat more distracting than conversational interference. Results using a babble of voices speaking Japanese were equivalent to results using an English babble. The conclusion is, therefore, that it is the spectrum of the masking signal and not its meaning which produces the observed interference with intelligibility.

8.5 ALTERNATING INTERFERENCES

In the course of the investigation of a wide variety of interfering signals, an adaptation effect was noticed for even experienced subjects when they were asked to listen for speech in the presence of some new and unfamiliar type of noise. Articulation scores rose as much as 30 per cent during the first 15 minutes of testing with a new signal. Apparently the adaptation effect results because the listener must decide what to listen to and what to ignore in the unfamiliar combination of speech and noise.

These observations suggested the possibility that masking might be greater if the masking signal were changed from moment to moment. If the masking noise were changed before the listener could become adapted to it, the masking efficiency of the alternating signals might be increased above the average effectiveness of the component signals. An experiment was designed, therefore, to discover the effect of alternating the interference at several fixed intervals and at random intervals.[3b] A slight effect in the expected direction was observed in the early stages of the experiment, but the effect disappeared as the listeners grew accustomed to the alternations. The maximum effect, obtained with a random rate of alternation, was a shift of about 5 per cent in the articulation score. The listeners reported that the alternating signals were much more distracting. In general, however, the effect of alternating the masking signals is negligible under the conditions of a laboratory test.

Chapter 9

THE INTERPHONE

IN MANY WAYS the interphone stands as the prototype for speech communication systems. The microphone, amplifier, and headset represent the indispensable elements: two electro-acoustic transducers joined by an intermediate transmitting network. The problems which arise are typical of the problems encountered in any communication system: intelligible transmission of speech under a variety of operating conditions, exclusion of noise at all points in the system, and the maximum possible comfort and convenience for personnel. Thus the following discussion of aircraft interphones exemplifies concepts and working principles which apply to a wide range of other equipment.

The aircraft interphone in use at the end of World War II represented the culmination of a series of modifications of the original equipment. Even before the entry of the United States into World War II, persistent reports of the unsatisfactory performance of aircraft interphones were received. A program of testing in the field and in the laboratory was begun, and improvements soon became available in microphones, headsets, and amplifiers. Changes were adopted, communication was improved, and at the close of hostilities every major component of the original interphone had been replaced.

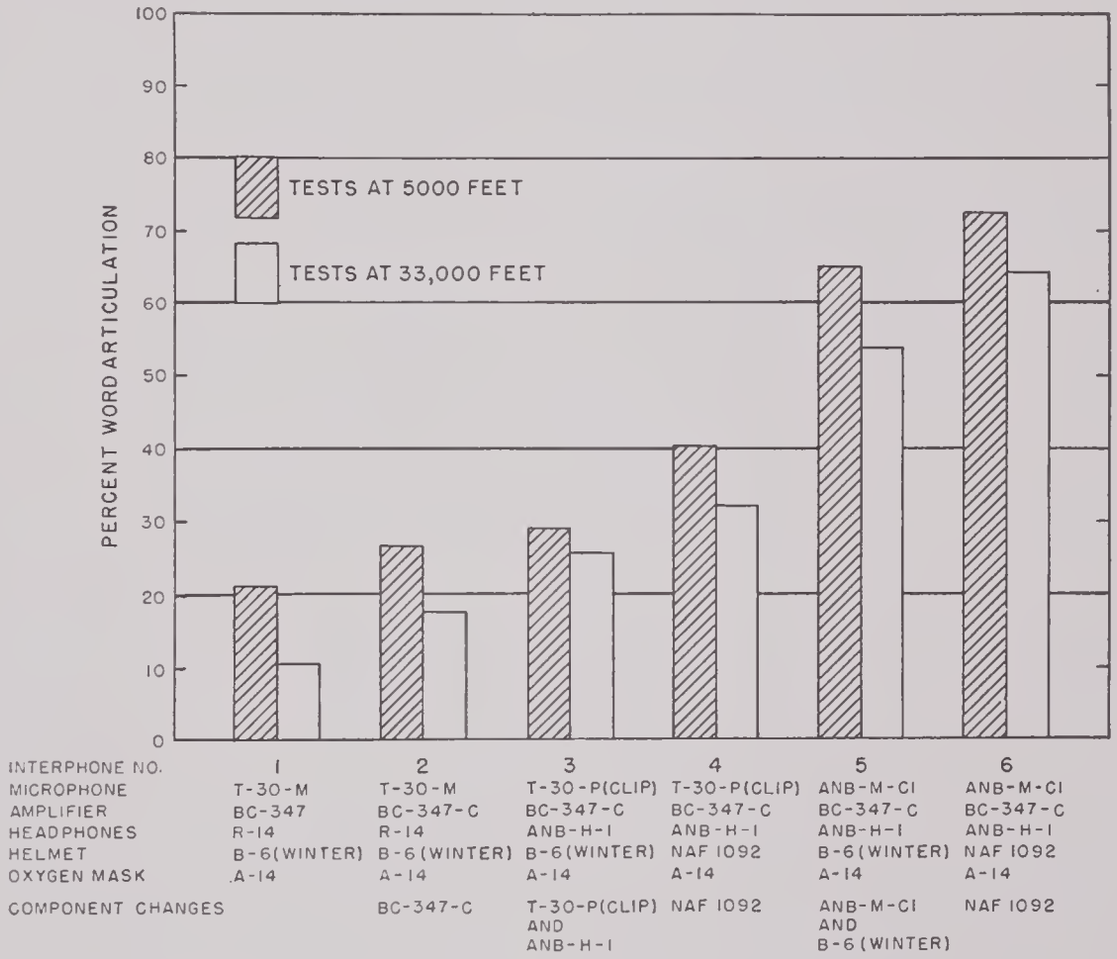


FIGURE 1. Bar graph illustrating the improvement in interphone performance as old components were discarded and replaced by newer ones.

The story of these improvements is graphically told in Figure 1. This chart shows the averaged results of a series of articulation tests on six aircraft interphones. The tests were run with trained Service personnel at low altitude

(hatched) and at high altitude (unhatched) during flights in a B-17F bomber at Eglin Field, Florida.[5] Interphone No. 1, at the left of the chart, represents approximately the performance of the original interphone. As the old components were discarded and replaced by newer ones, interphone performance improved steadily. The final version gave 60 to 70 per cent correct word articulation under operating conditions which had limited the earlier equipment to 10 or 20 per cent, and further improvements were still under development.

The value of these improvements can be measured in terms of greater military coordination. The personnel of a combat vehicle must communicate with one another to operate smoothly as a unit, and they must maintain contact with distant points of command to operate as an effective component of a larger team. The interphone is the standard system for communication between members of the crew, and the effectiveness of the vehicle depends in large measure upon the effectiveness of the interphone.

9.1 EVALUATION OF PERFORMANCE IN NOISE

One of the important contributions of the Psycho-Acoustic and Electro-Acoustic Laboratories in the early years of World War II was an insistence that the greatest single hazard to effective military communication is the presence of noise, and that interphone performance can be properly assessed only under the acoustic stress of ambient noise. Satisfactory performance by an interphone system in quiet surroundings does not insure satisfactory performance in noise. The evaluation of performance in noise became, therefore, a problem of major importance.

In its simplest form, the interphone operates as shown in Figure 2. When a member of the crew wants to talk to another station, he depresses his push-to-talk switch. The output voltage from his microphone is then amplified by the interphone amplifier and heard in the earphones at all stations. In such a system there are three major ways in which noise can reach the listener's ear (see Figure 3).

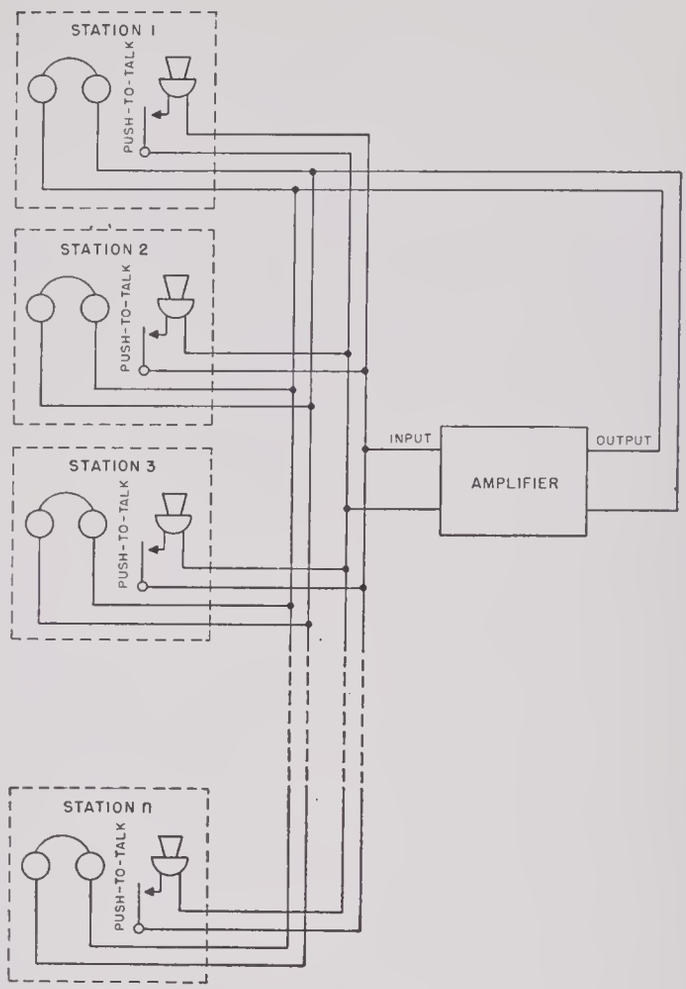


FIGURE 2. Interphone system (schematic).

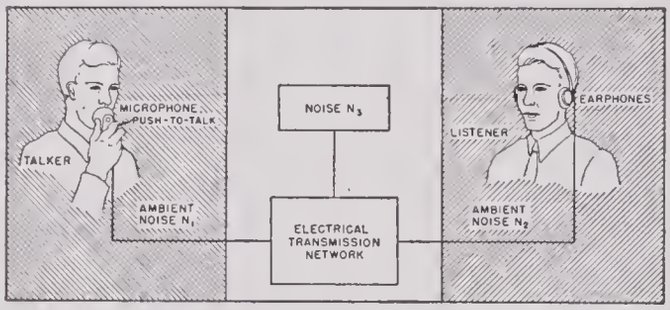


FIGURE 3. Noise interference in a voice communication system.

1. The microphone picks up a certain amount of the ambient noise N_1 around the talker. This noise reaches the listener via the interphone system. The noise spectrum reaching the listener may differ considerably from the spectrum of the ambient noise N_1. The noise spectrum acting upon the microphone is a function of the ambient sound field, of the type and shape of microphone, and of the position of the microphone relative to the mouth and face of the

talker. The spectrum is further modified before reaching the listener by the overall transmission characteristic of the interphone.

2. Noise may enter the listener's ear directly by leakage around the earphone and its cushion. This contribution depends on the noise spectrum N_2 at the listener's location and on the noise-excluding properties of the earphone cushion. These properties are a function of the type of earphone and cushion, and of the particular way in which they fit the listener's ear, and are independent of the rest of the system.

3. Noise may also enter the system by way of the electric transmission system itself. This source of noise, usually negligible for interphones, becomes important in other communication systems, e.g., a radio link. The noise spectrum at the listener's ear due to this cause is, in part, a function of the transmission characteristics of the system. Noise entering the system when the push-to-talk switch at other stations is accidentally pressed and noise picked up by the earphones acting as microphones at other stations can be lumped, if desired, with the effects of the noise N_3.

It is clear that an appraisal of the performance of interphones and their components must include not only their properties as electroacoustic transducers for the speech signal but also the degree to which unwanted sound is excluded.

9.1.1 Parameters Controlling Performance

The capacity of an interphone system[a] to transmit speech intelligibly is measured by a count of the number of discrete speech units recognized by the listener. The percentage of units correctly perceived is called the articulation score (see Chapter 5). The problem, therefore, is to correlate the efficiency of the interphone system, as measured by an articulation score, with its physical and psychophysical parameters.

Two theories relating the intelligibility of speech to the parameters of the interphone have been proposed for linear systems by the Bell Telephone Laboratories. According to one theory,[1, 9] the ability of the listener to perceive and recognize speech sounds depends *primarily* upon the relative intensities of the speech and noise spectra at the listener's ear, evaluated over most of the audible frequency range.

The spectrum level [SL] (see Section 4.1) of the received speech is a function of many variables, the most important of which are:

1. Voice level of the talker.
2. Orthotelephonic gain of the interphone (see Section 7.2.1).
3. Coupling between mouth and microphone.
4. Coupling between ear and earphone.
5. Speech material and composition.
6. Enunciation and training of the talker.

The spectrum level of the noise at the listener's ear depends on:

1. The ambient and electrical noise present.
2. The noise-excluding properties of the circuit, including microphone and earphone.
3. The efficiency and electrical properties of the circuit.

From this information and from certain assumptions and empirical relations derived from the fundamental properties of speech and hearing, an estimate can be made of the articulation scores to be expected. The estimates are most accurate when the spectrum of the noise at the listener's ear extends over a wide range of frequencies, the noise levels are moderate, and the response characteristic of the system is free from abrupt changes with frequency.

The correct calculation of scores on an absolute basis is difficult and of little practical meaning in view of the many variables involved. Comparison of measured and computed articulation scores on a relative basis, however, has yielded some useful results. A very desirable feature, from the design engineer's point of view, is that estimates can be made of the effects on the performance of the system of variations of circuit constants and elements. Predictions of performance based on this theory are considered in detail in Section 9.1.5.

The second theory[10] assumes that the differential sensitivity of the ear to frequency and level (see Section 3.5) is the controlling factor in the successful interpretation of speech sounds. Empirical functions were obtained at

[a] Talker and listener must be considered integral parts of a voice communication system.

the Bell Telephone Laboratories relating the speech spectrum, the differential sensitivity of the ear, the masking of sounds, and the articulation score. No use is made of this theory in the work reported here.

The important contribution of the two available theories is the demonstration that, in the simplest cases, the overall response of the system and its noise-excluding properties are of fundamental importance in determining performance. By assuming "average normal" talkers and listeners, standardized and uniform speech material, "constant" voice level, and negligible interaction between mouth and microphone, many of the troublesome variables can be effectively eliminated. However, this simplification, while useful for theoretical analysis, does not extend to many practical conditions of use. Hence, in evaluating interphone performance, articulation tests under simulated conditions of use should be performed wherever feasible.

9.1.2 Overall Response of the System

In the preceding section it was stated that systems can generally be rank-ordered as to performance under certain simplified conditions if the levels of received speech and noise at the listener's ear are known as a function of frequency. In order to obtain this information, it is necessary to consider the overall response of the system.

Meaningful comparisons of the *overall frequency response* of two systems can be most readily obtained by defining a suitable reference system. The orthotelephonic reference system (see Section 7.2.1) consists of a "normal" talker and listener[b] facing each other at a distance of 1 m in a quiet, nonreverberant room. The *orthotelephonic gain* [OG] at a given frequency is then expressed as the ratio of the level produced at the listener's ear by the system under test to that at the listener's ear when speech is transmitted from talker to listener over the 1-m path in free air. It is assumed, of course, that the talker speaks into the system under test in the same way he talks over the orthotelephonic reference system. By definition, then, speech heard over a system having an OG of 0 db at all frequencies will sound the same as speech heard over an air path 1 m in length.[c]

In order to obtain a quantitative measure of OG, it is necessary first to decide how the level at the listener's ear is to be measured. One method involves two measurements of the sound pressure at the listener's eardrum: first when he listens to the system under test and second when he listens to the talker over a direct air path 1 m in length. The ratio of these two pressures, expressed in decibels, is a direct measure of the orthotelephonic gain of the system.[14]

In practice, however, this measurement would be very difficult to perform. An alternative procedure, and one which yields the same results, involves the subjective loudness produced by a given sound source expressed in terms of the *equivalent free-field sound pressure*. The equivalent free-field pressure is obtained by requiring an observer located in a nonreverberant, quiet room to equate the loudness produced by a sound source some distance in front of him with that produced by the earphone. The equivalent free-field pressure is then defined as the free-field pressure, measured at the listener's location, which produces the same loudness as does the earphone under test. Specifically, by free-field sound pressure is meant here the pressure measured by a "point" microphone located at the position where the listener later places his head when making the loudness balance. This location is taken as the mid-point of a line joining the listener's ears.

The concept of equivalent free-field pressure, introduced by the Bell Telephone Laboratories, has the advantage that measurement of the sound pressure is more easily performed in the free field than at the listener's eardrum. Although there is some evidence to the contrary

[b] This is in contrast to methods using artificial voices and ears. The chief merit of using the latter lies in the simplicity achieved (see Chapter 10).

[c] If the experiment is actually performed in which a listener compares what he hears over a system with 0-db OG with what he hears over an air path of 1 m, certain differences, for example, in apparent localization may be experienced. Such differences are disregarded here.

(see Sections 3.2 and 10.3), it will be assumed here that for a given frequency and observer there is an essentially constant relation between the pressure at the eardrum and the corresponding equivalent free-field pressure for various types of earphones and pressure levels (see Section 3.1).

Consequently, the orthotelephonic gain of a system (in decibels) can be defined for any given frequency as follows:

$$\mathrm{OG} = 20 \log \frac{p_x}{p_0}. \qquad (1)$$

This equation can be expressed in words. The OG of a system X is the ratio, in decibels, of the equivalent free-field pressure p_x, produced by system X, to the free-field pressure p_0, generated 1 m from the talker in a quiet, nonreverberant room. Since the talker must speak the same way[d] into both system X and the orthotelephonic system, the OG of the system is independent of the intensity and spectrum of the talker's voice.

If the OG of a system is known as a function of frequency, the speech spectrum heard by the listener is easily obtained. First, the spectrum of the talker's voice is determined in terms of free-field pressure p_0, at 1 m in an anechoic (echo-free) chamber. Average speech spectra of this type were given in Figures 1 and 2 in Chapter 4. (They are easily reducible to a distance of 1 m.) Second, this speech spectrum is added (in terms of decibels) to the OG of the system X. The result is the speech spectrum heard by the listener over system X, expressed in terms of the equivalent free-field sound pressure p_x.

9.1.3 Synthesis of the Overall Response

In practice, it is convenient to synthesize the overall response of the system in three steps. The response of the three basic components of the interphone system—the microphone, the earphone, and the network connecting these two electro-acoustic transducers—is determined separately. These three response measurements, if properly made, can be added to obtain OG.

This procedure may best be illustrated by reconsidering equation (1) above. This equation can be rewritten in terms of three factors, as follows:

$$\mathrm{OG} = 20 \log \frac{p_x}{p_0} = 20 \log \left(\frac{e}{p_0}\right)\left(\frac{E}{e}\right)\left(\frac{p_x}{E}\right),$$

$$= 20 \log \frac{e}{p_0} + 20 \log \frac{E}{e} + 20 \log \frac{p_x}{E}. \qquad (2)$$

The three terms of equation (2) can be identified with the response characteristics of the three elements of the system in the following way.

Microphone. The term 20 log e/p_0 is the ratio, in decibels, of the voltage developed by the microphone, held in its normal relation to the talker, to the free-field sound pressure measured 1 m away from the talker with the microphone removed. An audio-spectrometer (see Chapter 4) is used to measure e and p_0 in suitable contiguous frequency bands. The measurement of p_0 is made under orthotelephonic reference conditions. The quantity 20 log e/p_0 may be referred to as the *real-voice response* of the microphone.

Amplifier. The term 20 log E/e is the ratio, in decibels, of the voltage E developed across the listener's headset to the voltage e developed by the microphone. This term is simply the electrical response of the network connecting the two electro-acoustic transducers.

Earphone. The term 20 log p_x/E is the ratio of the equivalent free-field sound pressure p_x to the voltage E measured across the terminals of the earphone. More specifically, p_x is the free-field pressure necessary to produce the same loudness sensation as that produced when a voltage E is applied to the earphone. In making the loudness-balance judgment the listener is located in a quiet, nonreverberant room with the sound source in front of him. It is not necessary to use a speech signal. The quantity 20 log p_x/E may be referred to as the *real-ear response* of the earphone.

Addition of these three terms gives the orthotelephonic gain for the system at any given frequency.

[d] Difficulties in keeping the voice "constant" are encountered if marked interaction exists between the talker and the microphone he uses, as, for example, when an oxygen mask is used.

9.1.4 Response of the System to Noise

The level and spectrum of the noise at the listener's ear is the sum of the contributions from the noise transmitted by the microphone, the noise reaching the listener's ear through lack of insulation of the earphone cushion, and any contributions from noise due to other causes (see Figure 3). For noise of the type found in combat vehicles (see Chapter 2), the resultant noise spectrum at the listener's ear is obtained by summation (on a power basis) of the various contributions.

The noise entering the system through the microphone can be measured readily by using an audio-spectrometer (see Section 10.1.2). A "talker" (who, however, does not talk) holds or wears the microphone under test in a diffuse sound field corresponding to N_1 (see Figure 3), while the voltage developed by the microphone due to noise pickup is measured. By addition of the amplifier response and real-ear response, the equivalent free-field spectrum level of the noise due to N_1 is obtained.

To obtain the noise spectrum leaking through the earphone cushion, the listener might be immersed in the diffuse noise field N_2 wearing the earphones under test in normal conditions of use, and a measurement made of the noise spectrum at the eardrum would yield the desired results. A much simpler method determines the attenuation which the headset provides for pure tones (see Section 10.3.2). The attenuation, in decibels, is then subtracted from the noise spectrum N_2, and the spectrum reaching the listener's ear through the cushion is thus expressed approximately in terms of equivalent free-field sound pressures.

Still another method consists in measuring the masked threshold of the listener for pure tones when noise is leaking through the cushion. This is accomplished by energizing the earphone with single frequencies. This method is, in many ways, the most fundamental one, in that it gives a direct measure of the reduced auditory area available for communication. It takes into account such effects as masking by adjacent noise components (spread of masking). It can be used to test the noise exclusion properties of earphones (see Section 10.3.2), but is an even more valuable laboratory method if applied to the *total* noise spectrum at the listener's ear due to N_1, N_2, and N_3.

The contribution due to electrical noise N_3 is evaluated by an analysis of the voltage appearing across the earphone terminals. Addition of this spectrum to the real-ear response of the earphone yields the desired result.

9.1.5 Prediction of Performance

Now that we have considered in detail the procedures for determining the speech and noise spectra at the listener's ear, we can proceed to demonstrate how this information is used to predict the performance of an interphone system. The following computational procedures were devised originally by the Bell Telephone Laboratories[14a] and later refined in some respects by the Electro-Acoustic Laboratory.[14a]

It is the fundamental assumption of the theory that the contribution to intelligibility by any given band of speech frequencies may be calculated independent of the contributions made by other bands. These contributions can be expressed by means of an empirical function, the *articulation index* [AI], which has additive properties.[e] The total contribution to AI over a total range of frequencies is the sum of the contributions to AI of the component bands contained in that range, essentially independent of the manner of subdivision.

The sum of the contributions of all the bands of frequencies passed by the interphone can be written as follows:

$$\mathrm{AI} = \sum_{p=1}^{p=n} (\Delta \mathrm{AI})_p, \tag{3}$$

where $(\Delta \mathrm{AI})_p$ is the contribution to the articulation index of the pth band of speech frequencies in the total range under consideration. Clearly, $(\Delta \mathrm{AI})_p$ is a function of both the frequency limits of the band and of the speech and noise levels in that band.

[e] It is also assumed that there is a unique relation between articulation score and AI for a given speech material and crew.

The problem, of course, is to design the interphone in such a way as to maximize AI, i.e., to maximize the contribution from each of the p bands of frequencies. It is plausible to assume that this condition will obtain for low levels of noise and high levels of speech, and $(\Delta AI)_p^{max}$ will then depend primarily upon frequency.

We therefore put

$$(\Delta AI)_p = (\Delta AI)_p^{max} W_p. \quad (4)$$

The weighting factor W_p can be interpreted as the fractional part of $(\Delta AI)_p^{max}$ which is available when the speech and noise levels are unfavorable. Thus W_p is primarily a function of the speech-to-noise ratio in the pth band, and it is convenient to assume that it is independent of frequency. Hence,

$$AI = \sum_{p=1}^{p=n} (\Delta AI)_p^{max} W_p. \quad (5)$$

It is necessary, therefore, to evaluate $(\Delta AI)_p^{max}$ experimentally as a function of frequency and W_p as a function of the signal-to-noise ratio.

Relation between $(\Delta AI)_p^{max}$ and Frequency

This relation has been evaluated at the Bell Telephone Laboratories for nonsense syllables

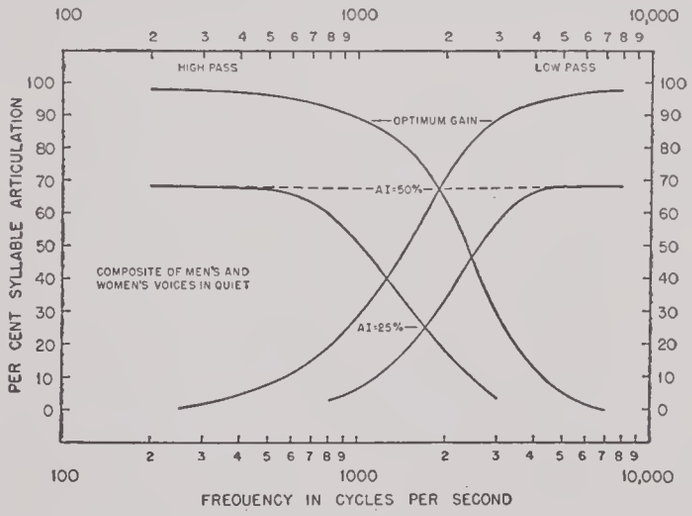


Figure 4. Articulation scores vs cutoff frequency of high-pass and low-pass filters inserted in a wide-band system. (BTL)

with both men and women talking. Articulation tests were made in quiet on a high-fidelity system into which low-pass and high-pass filters could be inserted. The results of these tests are presented in Figure 17 of Chapter 7.

If the optimal scores for each filter condition are plotted as a function of filter-cutoff frequency, the upper curves in Figure 4 result. It is evident from these curves that as much of the total intelligibility is carried by frequencies below 1,900 c as is carried by the frequencies above 1,900 c. Furthermore, the first point on the additive AI scale, the 50 per cent point, is thus determined. It appears that 50 per cent AI corresponds to an articulation score of 68 per cent. Another point can be obtained by returning to the orthotelephonic gain of the wide-band system which gives an articulation score of 68 per cent. From Figure 17, Chapter 7, we see that the wide-band system gives 68 per cent articulation with an orthotelephonic gain of about —31 db. If the articulation scores are now plotted as a function of filter-cutoff frequencies for this gain setting, the lower curves

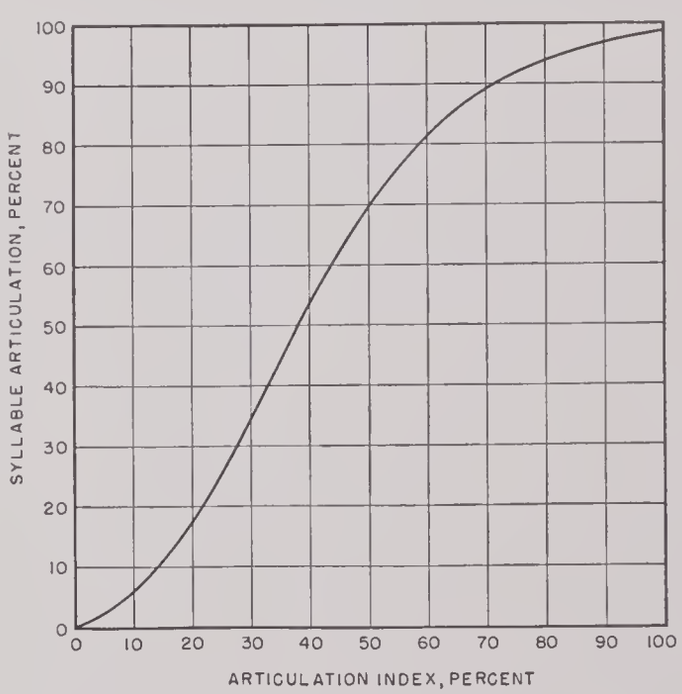


Figure 5. Approximate relation between syllable articulation and articulation index. (BTL)

in Figure 4 result. These functions cross at a point which defines an AI of 25 per cent.

By a similar fractionating procedure, the complete function relating AI to syllable articulation score can be plotted. The function has the general shape shown in Figure 5. For different speech material and crews, other curves would

have been obtained. By means of this function, the ordinate in Figure 4 can be converted directly into AI. This has been done in Figure 6, where the cumulative contribution to the articulation index is shown as a function of the upper cutoff frequency. The slope of this function shows the relative importance of the various frequencies in their contribution to intelligibility. The similarity between this function (see Figure 6), the pitch scale (see Figure 7, Chapter 3), and the function which relates the position of maximum agitation on the basilar membrane to the frequency of the acoustic stimulus has been pointed out (see Section 3.3.1).

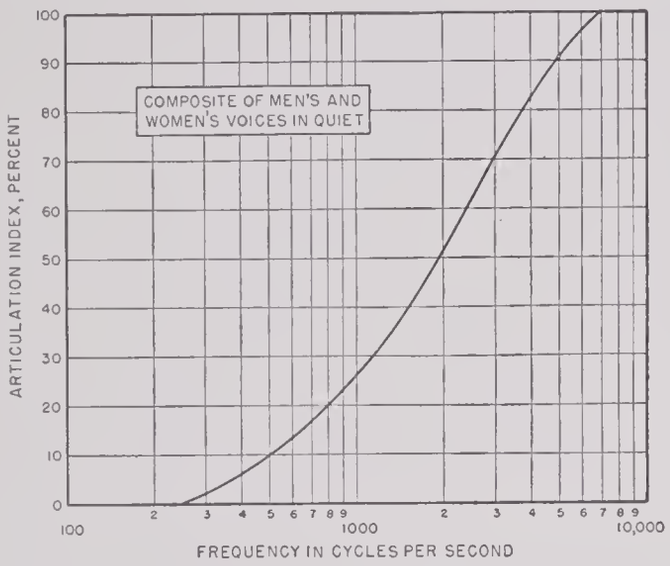


FIGURE 6. Relating articulation index and cut-off frequency of a low-pass filter inserted in a wide-band system. (BTL)

With this information, the range of important speech frequencies can be divided into a number of bands which contribute equally to the articulation index. Computations are normally made on the basis of 20 bands, each contributing 5 per cent.

RELATION BETWEEN W_p AND THE LEVELS OF SPEECH AND NOISE

As the simplest case, consider first the computation of W_p for quiet conditions. If the entire frequency range of speech is divided into 20 bands, each contributing 5 per cent to AI, then $(\Delta AI)_p$ can be plotted between 0 and 5 per cent as a function of the orthotelephonic gain for each of the 20 bands. When this is done, it is found that W_p depends principally upon the sensation level of the speech in the band, i.e., upon the amount by which the speech level exceeds the threshold of hearing in the band.

These results can be carried over directly to the case where noise is present if it is assumed that we are dealing with the masked threshold instead of the quiet threshold of hearing. Now we have previously seen (see Section 3.4.1) that the masking of pure tones by noise is directly proportional to the effective level Z of the band of noise frequencies (see Figure 10, Chapter 3). Hence W_p can be expressed directly in terms of the speech-to-noise ratio in each band. The spectrum level of the noise B in decibels is subtracted from the spectrum level of the speech B_s in decibels. Because the temporal distribution of amplitudes differs for speech and noise, a constant number of decibels is added to the speech-to-noise ratio $(B_s - B)$ in order to fit the experimental evidence.[1b]

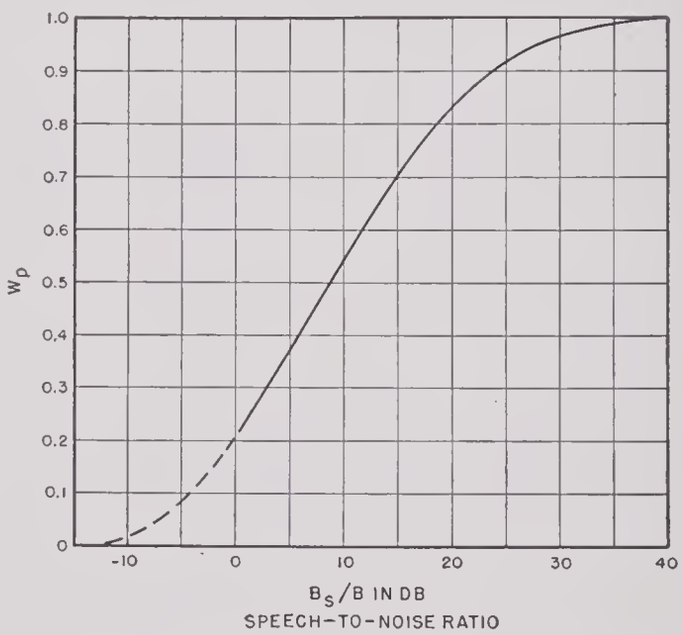


FIGURE 7. Weighting factor vs speech-to-noise ratio. (BTL)

Figure 7 shows approximately how the weighting factor W_p depends upon the speech-to-noise ratio $B_s - B$. If the signal-to-noise ratios have been determined for each of the 20 bands of speech frequencies, the corresponding values for W_p can be read from Figure 7. It can be seen that the speech-to-noise ratio must be of

the order of 20 to 25 db before a band of speech frequencies makes its maximum contribution to the articulation index.

In making calculations, a work sheet like the one shown in Figure 8 may be helpful. The frequency scale is divided into 20 bands, each having a maximum potential contribution to

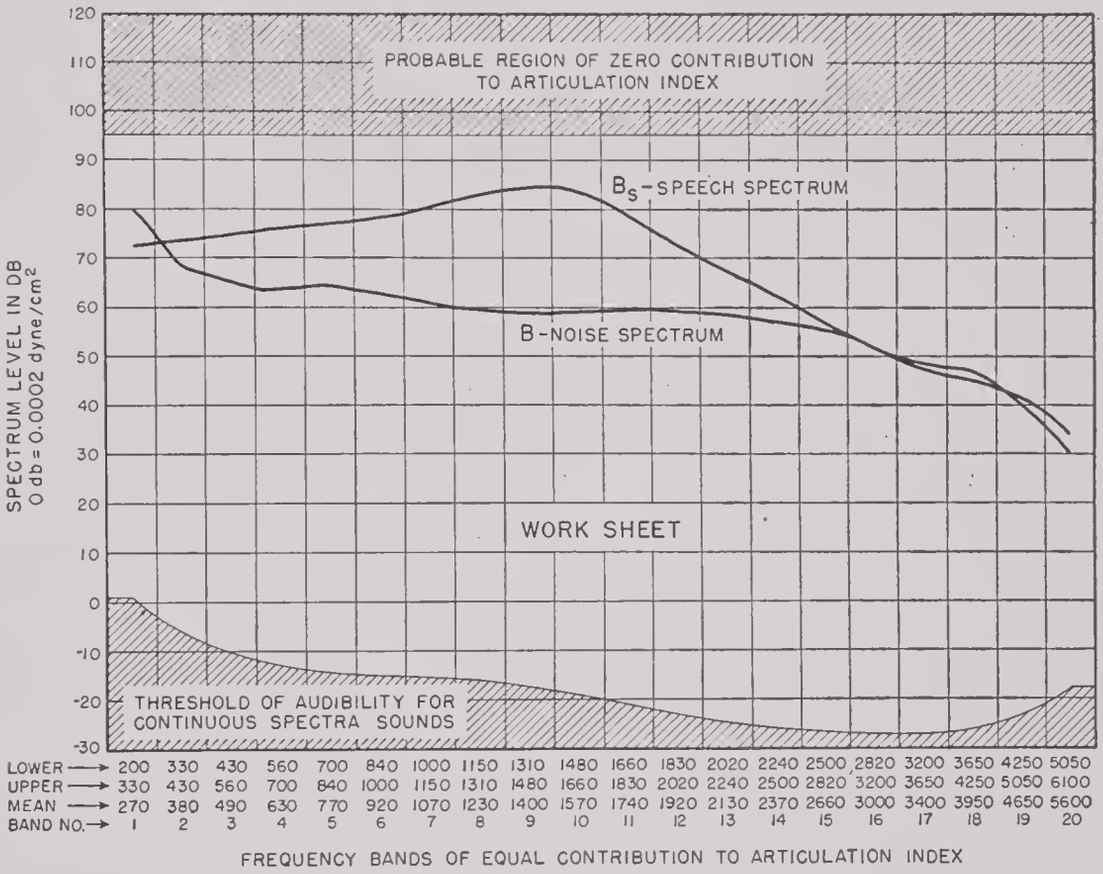


FIGURE 8. Work sheet for evaluating the articulation index of a communication system with noise present.

AI of 5 per cent. The weighting factor can then be obtained from Figure 7 for each band. Addition of all contributions to AI according to equation (5) determines the total articulation index. If it is desired, once AI is known, the syllable articulation score can be predicted from Figure 5. In this manner, an approximation of the interphone's performance is obtained solely on the basis of the speech-to-noise ratio at the listener's ear.[f]

Figure 9 shows a comparison of theory and experiment. The system under test was a band-pass system of variable bandwidth. The noise spectrum at the listener's ear is shown in Figure 10, expressed in terms of the equivalent free-field spectrum level. While, in some instances, the agreement between the experiment and the calculations is not especially close, the theory seems to provide a workable estimate of the relative performance of the systems.

[f] Refinements have been added to the theory, taking into account such secondary variables as interband masking, self-masking, and nonlinear distortion of certain types.

9.2 INTERPHONE COMPONENTS: THE MICROPHONE

The task of delivering a signal at a suitable speech-to-noise ratio over a wide band of frequencies is the joint responsibility of the microphone, the amplifier, and the headset. Failure on the part of any one of these components means failure of the entire interphone.

The first link in the chain is, of course, the microphone. Several different types of microphones were used in World War II: hand-held microphones, noise-canceling microphones, throat microphones, microphones mounted in enclosures. Each type was represented by a variety of models, and it is necessary, therefore, to limit the following discussion to those micro-

phones which were actually used on a large scale and to the general factors governing the performance of each of the generic types.

9.2.1 Hand-Held Microphones

The two hand-held microphones most widely used in aircraft were the Army's T-17 and the Navy's T-38 (RS-38-A). Photographs of these

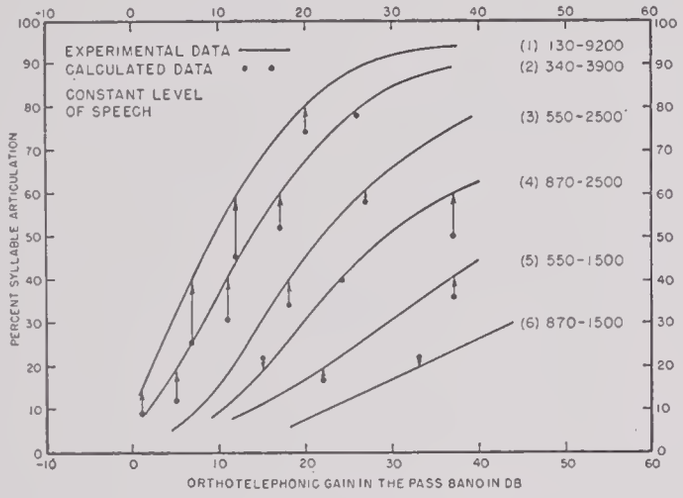


FIGURE 9. Measured and computed gain functions for band-pass systems in noise.

two microphones are shown in Figures 11 and 12. Both the T-17 and the T-38 are carbon-button instruments.

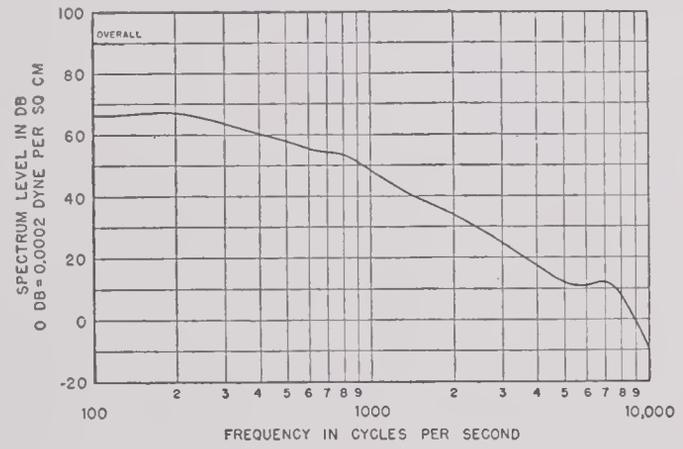


FIGURE 10. Noise spectrum at the listener's ear in articulation tests described by Figure 9.

Carbon-type microphones were adopted almost exclusively by the Armed Forces of the United States, despite the fact that all carbon microphones are essentially nonlinear devices. Furthermore, their frequency-response characteristic depends upon the level of the sound pressure for which the response is obtained. They require a separate source of current, their performance is likely to vary with change of position, and they produce a certain amount of "burning noise" due to the passage of current through the carbon granules. In addition, there is some deterioration of the carbon granules with age. Their extensive use is due primarily to their very high sensitivity, far exceeding the sensitivity of magnetic and dynamic microphones. In spite of their limitations, the T-17 and T-38 microphones proved to be reliable instruments for communication.

Hand-held microphones as a class have certain disadvantages which limit their usefulness. They do not exclude ambient noise as well as the other types of microphones, and they require the continuous use of one hand during communication.

Measurements of the speech-to-noise ratio were made for these and other microphones. Eleven sets of highly discriminative band-pass filters, having cutoff frequencies indicated on the figures, were used for the analyses. A talker held the microphone close to his mouth in an ambient noise field similar to that found in two-engined aircraft without sound treatment. The talker remained silent while the noise level from the microphone was measured for each of the pass bands. The noise was then turned off, and the analysis repeated as the talker spoke the sentence, "Joe took father's shoe-bench out; she was waiting at my lawn." The results of these measurements are presented in Figures 13 and 14. The speech-to-noise ratio in any band is found by counting the number of decibels between the levels of the two signals.

Measurements of this type are influenced by a variety of variables and therefore should never be regarded as absolute. Their value lies in the fact that they provide a basis for comparing different microphones all tested under identical conditions. Comparison of the T-17 and T-38 shows that the speech-to-noise ratios are nearly the same at all frequencies. This agrees with the usual finding that the two microphones yield approximately equivalent articulation scores when tested under the same conditions.

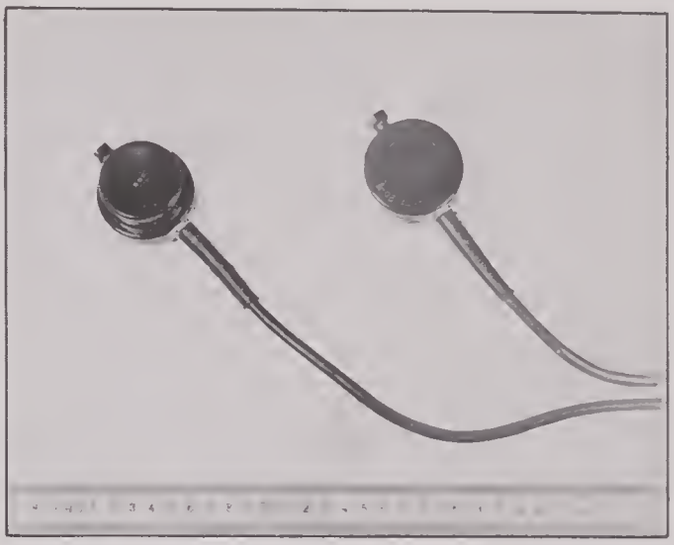


FIGURE 11. Hand-held carbon microphone T-17.

FIGURE 12. Hand-held carbon microphone T-38.

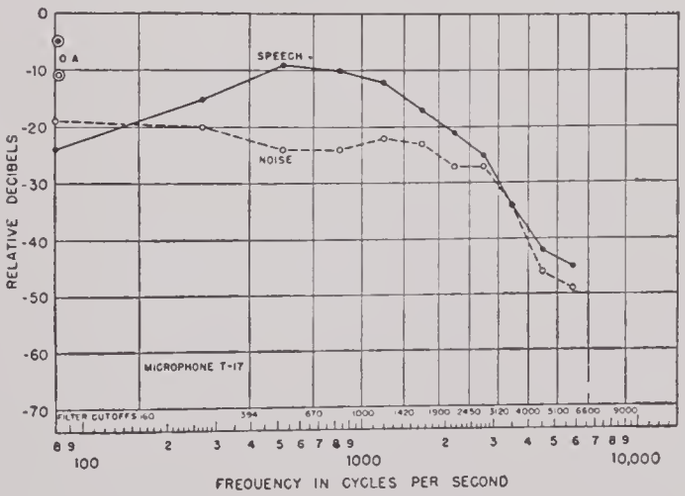


FIGURE 13. Speech and noise levels obtained for different frequency bands with the T-17 microphone.

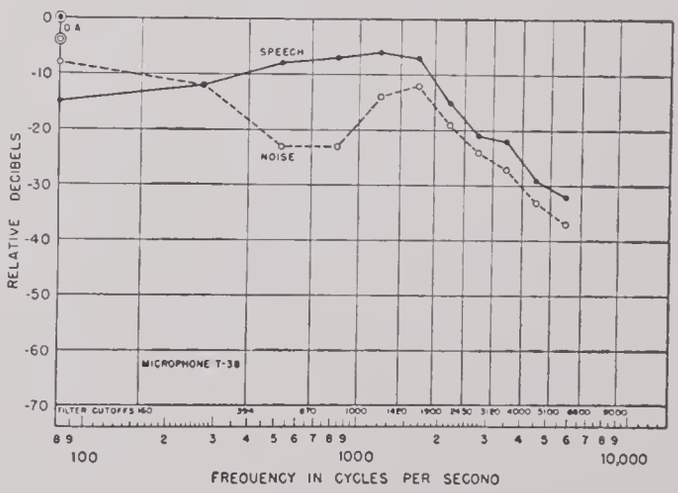


FIGURE 14. Speech and noise levels obtained for different frequency bands with the T-38 microphone.

9.2.2 Noise-Canceling Microphones

Noise-canceling, or differential, microphones have both sides of the diaphragm exposed to the impinging sound waves and are, therefore, sensitive to the pressure gradient in the sound field. When the differential microphone is close to and oriented toward the talker's lips, it is activated primarily by the talker's voice and not by the noise field.

FIGURE 15. Noise-canceling T-45 microphone.

The noise-canceling microphone, T-45, used by the Army Ground Forces, is shown in Figure 15. A later improved version, M-5/UR, is shown in Figure 16. Because of the manner of suspension in front of the talker's lips, these were sometimes called "moustache" or "lip" microphones. This suspension leaves the hands free (if a suitable switch is provided), and thus has one great advantage over the hand-held microphone.

Speech and noise measurements for noise-canceling microphones, comparable to those presented for hand-held microphones, are shown in Figures 17 and 18. It can be seen from these figures that the speech-to-noise ratio is most favorable at frequencies below about 2,000 c. In this region the ratio is somewhat better than that of the hand-held microphone, although neither microphone performs very poorly here. Unfortunately, the noise-canceling microphone possesses an inherent limitation in its inability to discriminate against high-frequency noise, and in high ambient noise this

FIGURE 16. Noise-canceling M-5/UR microphone.

becomes a serious consideration.[13] It is usually in the region above 3,000 c that noise levels actually exceed speech levels, and in this region the noise canceler is just another microphone.

Figures 17 and 18 also include measurements of "wind noise." The talker wore the microphone in a 30-mph wind and adjusted the position of his head until the maximum level of noise was obtained. He then held this position, and the noise due to the wind was measured in each frequency band. The results show that the T-45 is more susceptible to wind noise than the M-5/UR, and this was confirmed by the results of articulation tests. In both cases,

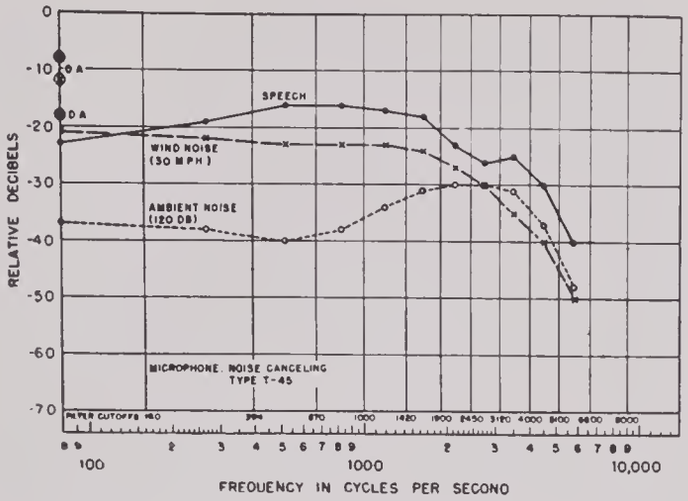


FIGURE 17. Speech and noise levels obtained for different frequency bands with the T-45 microphone.

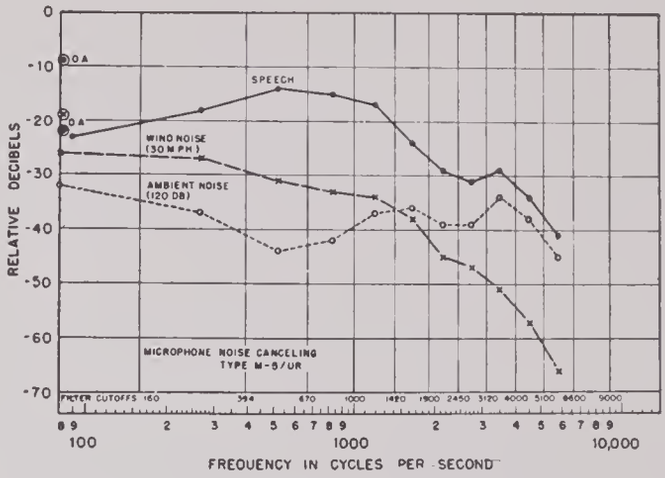
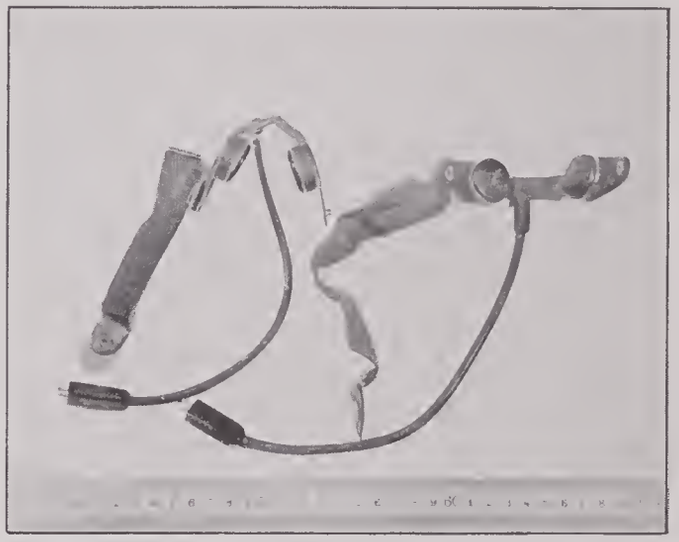


FIGURE 18. Speech and noise levels obtained for different frequency bands with the M-5/UR microphone.

however, the noise-canceling microphones generate a higher overall noise level in this wind than they do when used in an ambient noise of 120 db.

While the noise-canceling microphones possess limitations which cannot be overcome, they proved to be the most satisfactory devices available in moderate or low noise levels or in instances where acoustic feedback was a critical problem. It was a real misfortune that the sensitivity of aviators' lips, or their fondness for moustaches, and the relative lack of sensitivity of their necks should have delayed the substitution of this microphone for the ubiquitous throat microphone.



9.2.3 Throat Microphones

A device used widely by the U. S. Army Air Forces at the beginning of World War II was the throat microphone. In this assembly the microphone is strapped to the throat directly above the larynx. Such an arrangement possessed the advantages of apparently low noise pickup and free use of hands, and it would probably have been a very effective instrument but for the fact that the speech signal available at the larynx is intrinsically unintelligible. A photograph of the standard throat microphone, T-30, is shown in Figure 19.

FIGURE 19. Throat microphone T-30.

Articulation tests showed persistently that the "mushy" speech picked up by a throat microphone provides a poor means of com-

munication. That this inferiority was not due to poor design of the American version is illustrated by the articulation results shown in Figures 20 and 21. These figures present the

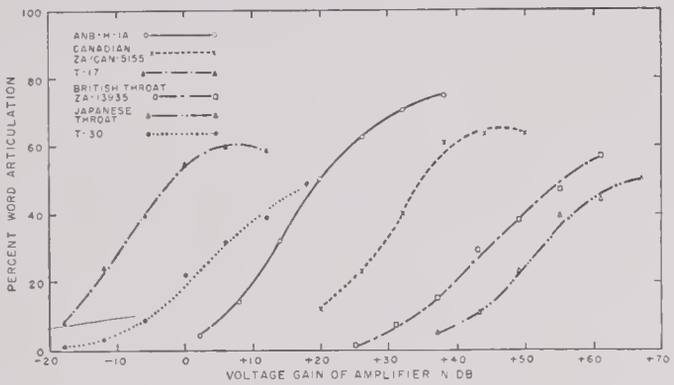


FIGURE 20. Articulation as a function of the voltage amplification applied to the outputs of six different microphones.

results of articulation tests run in 120 db of airplane noise with three hand-held and three throat microphones.[11] The Japanese and the British (ZA 13935) throat microphones are both magnetic instruments; the American T-30 is of the carbon type. The hand-held microphone was represented by the American T-17. The measurements included the Canadian dynamic ZA/CAN 5155 in a noise shield, and a special instrument consisting of the dynamic earphone ANB-H-1A used as a microphone. Figure 20 shows the articulation of each of these microphones plotted as a function of the voltage gain of the interphone amplifier. The two carbon microphones, T-17 and T-30, are clearly the most sensitive and require the least amplification, while the two magnetic instruments, the British and Japanese throat microphones, are the least sensitive. If these functions are now replotted according to the level of the speech received at the listener's ears, they group as shown in Figure 21. The three hand-held microphones are clearly superior to the three throat microphones.

This inferiority of the throat microphone was repeatedly demonstrated by tests in the laboratory and in the field, in spite of the fact that the overall noise levels seemed favorable. Figure 22 shows the speech and noise levels comparable to those presented above. It will be noted that the speech level falls rapidly above 1,500 c and this correlates with the "mushy,"

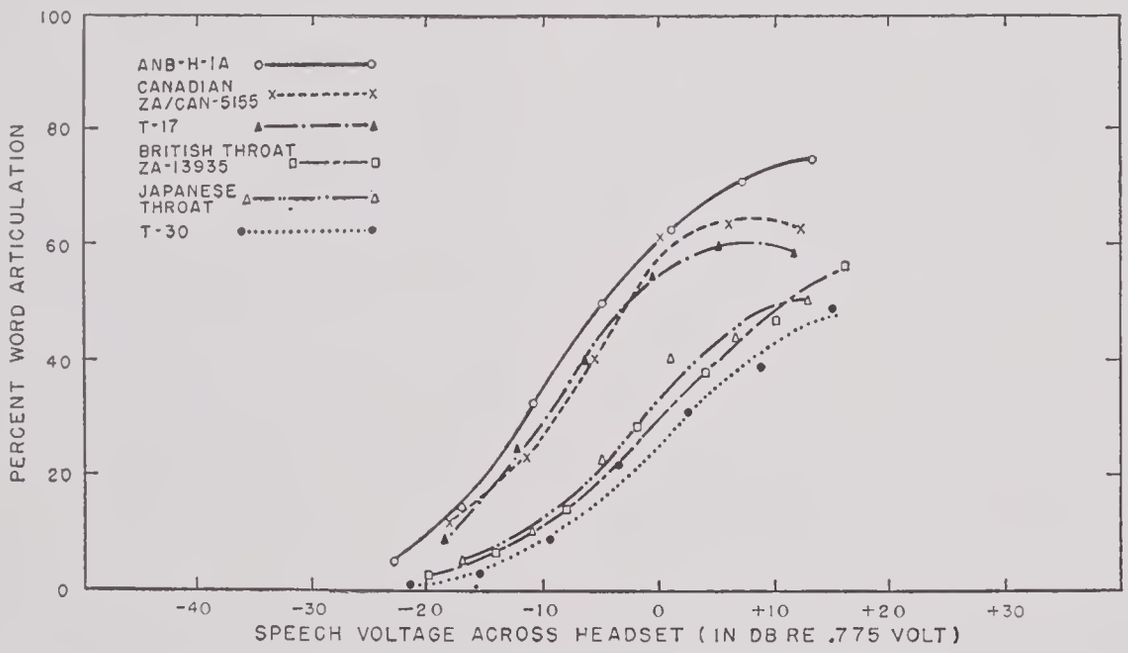


FIGURE 21. Word articulation provided by six microphones when the amplification used with each microphone was adjusted to produce approximately the same speech level at the listeners' ears. Each point is based on 200 words.

consonantless quality of the signal. In this instance, therefore, the usual advantage of a positive speech-to-noise ratio is of little value because the speech itself is so unnatural.

The second merit of the throat microphone,

the freedom of the hands, also proved illusory in practice. The position of the microphone was critical and required frequent adjustment, and talkers who were frustrated by the poor performance of the instrument would often use

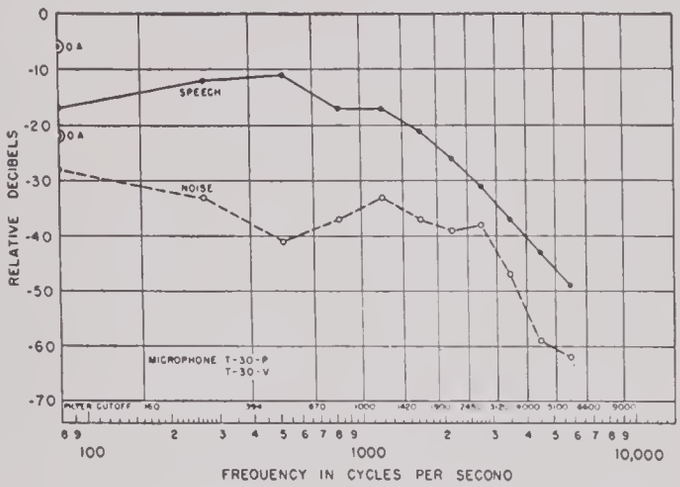


FIGURE 22. Speech and noise levels obtained for different frequency bands with the T-30 microphone.

one hand to press the microphone against their throats. All in all, the results with the throat microphone were not what had been hoped, and its belated replacement by other types of microphones was one of the major improvements made in the interphone system.

9.2.4 Microphones Mounted in an Enclosure

Mounting a small microphone inside an oxygen mask or a noise shield combines the advantages of noise exclusion with free use of the hands. The effect of such an arrangement, however, is to reinforce the low frequencies, and the speech acquires an unfortunate, "booming" character. One of the fundamental requirements, therefore, is that the mask microphone should have a poor low-frequency response in order to make the response of the mask and microphone combination approximately uniform. The fidelity and level of speech transmitted over these devices depends to a large extent upon the acoustic properties of the enclosure, and the interaction between the response characteristics of the mask and the microphone poses a very difficult problem for analysis. In most cases an empirical evaluation is necessary.

The mask microphone adopted by both the Army and Navy was a carbon instrument, ANB-M-C1. A magnetic mask microphone, MC-253-A, was tested extensively and was adopted for use by the British. A photograph of these two microphones is shown in Figure 23.

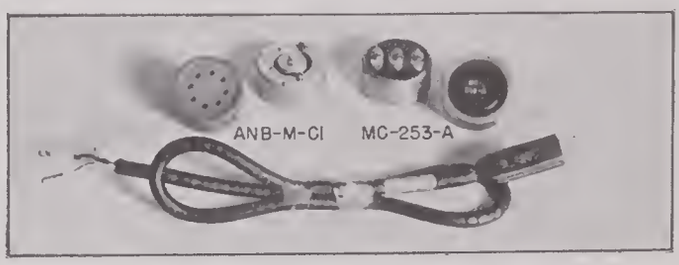


FIGURE 23. Mask microphones ANB-M-C1 (carbon) and MC-253-A (magnetic).

As in the case of the throat microphone, the interaction between the mask microphone and the voice precludes the simplifying theoretical assumptions made in Section 9.1.5. The articulation efficiency which can be realized with a properly designed mask microphone is considerably better than that obtained with throat microphones. When oxygen masks must be used, therefore, it is recommended that the microphone be placed inside the mask rather than on the throat. A photograph of the standard oxygen mask, A-14, used by both the Army and Navy, is shown in Figure 24. The microphone is mounted in the mask just above the outlet valve.

There are also some advantages to be gained at low altitudes by mounting the microphone in an enclosure. Noise shields were developed which provided a good speech-to-noise ratio, eliminated wind noise, and left the hands free. A photograph of the ANB-M-C1 microphone mounted in the Harvard D-17 noise shield is shown in Figure 25. Speech, ambient noise, and wind noise measurements are shown in Figure 26. The speech-to-noise ratio is excellent at all frequencies, and the effect of a 30-mph wind is much less serious than in the case of the noise-canceling microphones (see Figures 17 and 18). A noise shield was also developed by the Bell Telephone Laboratories which covered the nose and the mouth and which performed even better than the Harvard D-17 shield.

Neither of these shields was adopted by the Services, however.

An interesting comparison of the four types of microphones is given in Figure 27.[7] Articulation tests were run in ambient noise similar to that encountered in the cabin of a two-engined aircraft without sound treatment. Speech levels were held constant throughout all the tests. Five of the six microphones were nearly equivalent; the throat microphone T-30 was strikingly inferior. Best results were obtained with the ANB-M-C1 in a noise shield, but the superiority over the noise-canceling microphones was small. The hand-held T-17 ranks fifth in this group.

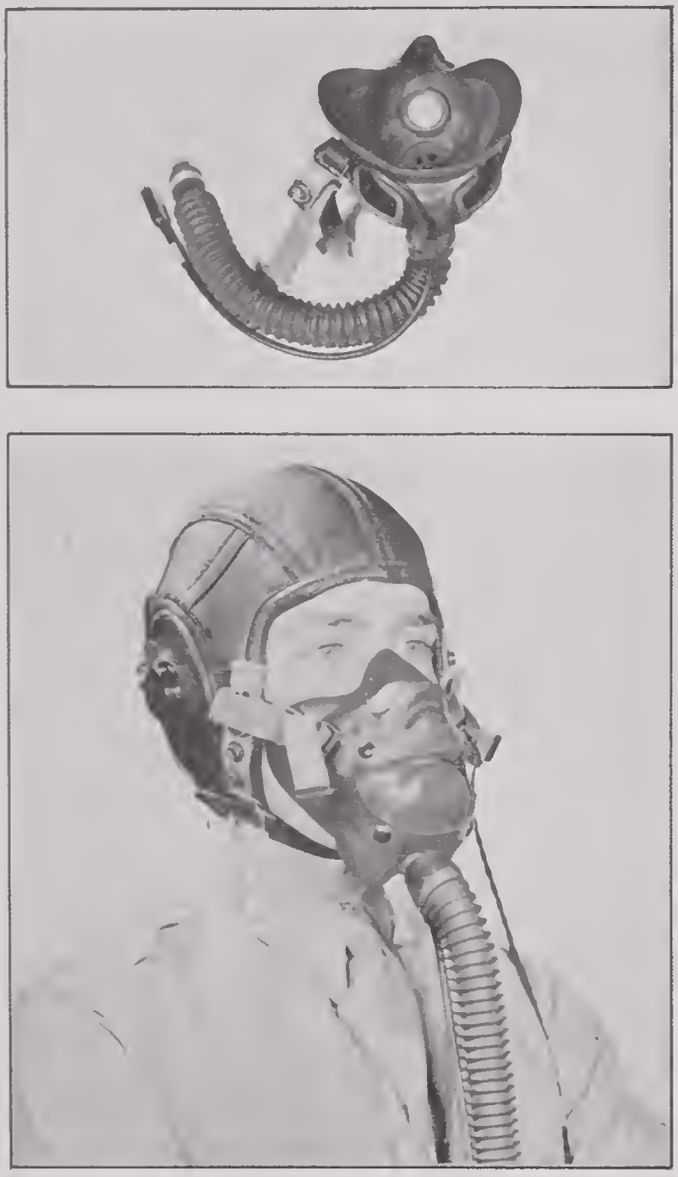


FIGURE 24. Oxygen mask A-14.

From such comparisons it appears that the noise shield has much to recommend it. Noise shields are not, however, as comfortable as some of the other classes of microphones and, in certain situations, they might restrict the useful field of vision. Also, they would have to be issued individually, whereas the noise-canceling microphone does not. These considerations probably contributed to the adoption of noise-canceling microphones rather than noise shields for use at low altitudes.

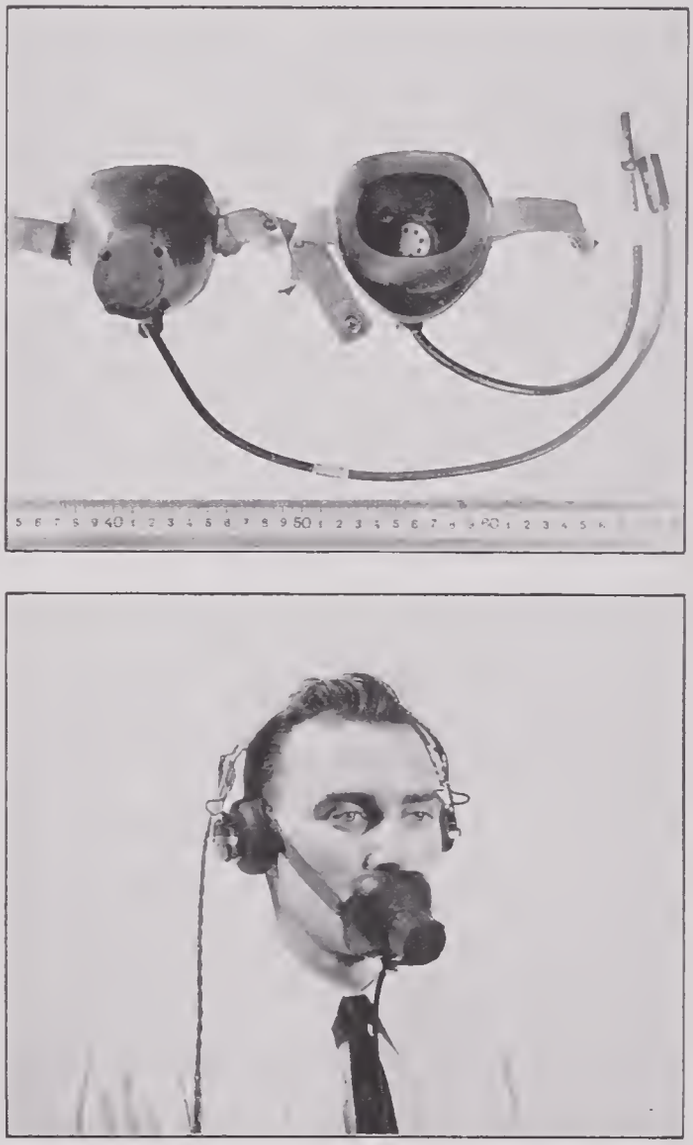


FIGURE 25. Noise shield D-17.

9.3 INTERPHONE COMPONENTS: THE AMPLIFIER

The second link in the chain from talker to listener is the interphone amplifier. The requirements for a satisfactory amplifier are

straightforward and can be easily met by careful engineering: uniform response, sufficient voltage gain, and a power output adequate for the number of headsets used. The problem of

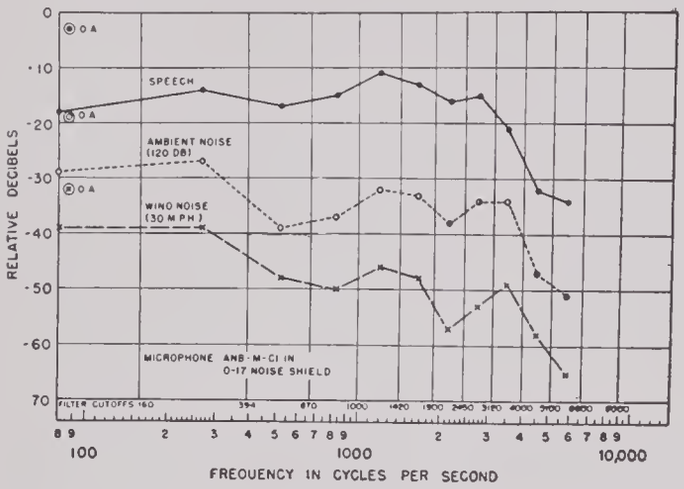


FIGURE 26. Speech and noise levels obtained for different frequency bands with the ANB-M-C1 in the D-17 noise shield.

noise exclusion, so serious for microphones and headsets, is not a factor in the design of the amplifier.

The interphone amplifier BC-212, with which

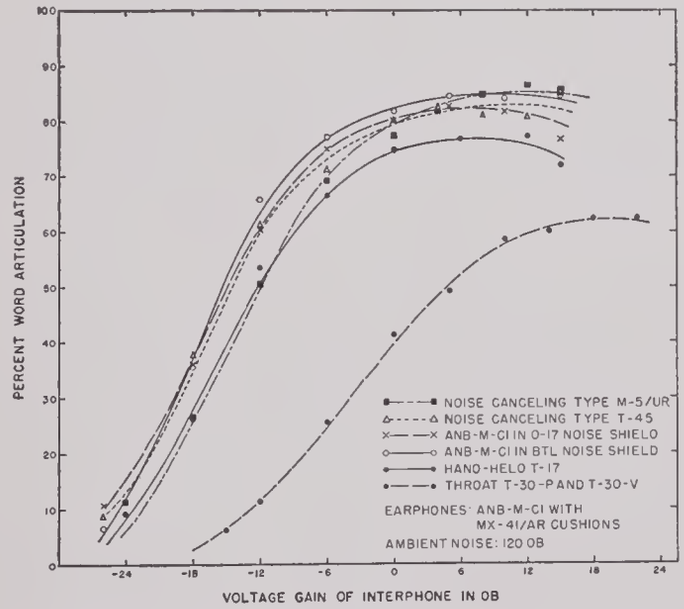


FIGURE 27. Word articulation as a function of voltage gain for six carbon microphones.

the AAF began World War II, satisfied none of the fundamental requirements for satisfactory performance, and it was soon replaced by a more adequate unit, BC-347C. This amplifier possessed a satisfactory frequency response and was fairly adequate for communication at low altitudes. At high altitudes (see Section 9.5) there is a marked decrease in headphone sensitivity and in the voice level which a talker is able to maintain. Extensive flight tests at Eglin Field[6] showed that at altitudes of 35,000 ft neither the voltage nor the power-output capacity of the amplifier BC-347C were adequate. As a result of these tests, design recommendations were formulated. The frequency response at sea level should be essentially flat between 300 and 4,000 c, with an overall voltage gain of approximately 18 db. At 35,000 ft the overall gain should be increased to approximately 30 db, and the amplifier should be capable of delivering a peak power of at least 200 milliwatts per headset, preferably 400.

Following these recommendations, the interphone amplifier AM-26/AIC was adopted by the Army Air Forces. This amplifier possesses a power-output capability of approximately 5 watts as compared to approximately 0.75 watt available from the amplifier BC-347C which it replaced. In addition, the AM-26/AIC provides increasing voltage gain with altitude.

The improvements in the amplifier, therefore, were all changes in the direction of more uniform frequency response and more adequate output power. Once the minimum requirements had been clearly demonstrated by experimental tests, most of the difficulties were readily overcome.

9.4 INTERPHONE COMPONENTS: THE HEADSET

The headset consists of three components: the earphones, the earphone sockets, and the supporting headband, neckband, or aviation helmet.[g] Its performance depends in part upon the functioning of each of the components separately and in part upon the way in which they interact upon each other.

The basic function of the earphone is to convert electric into acoustic power. Its effectiveness depends primarily upon the characteristics of the earphone itself, upon the dimensions and

[g] For present purposes the aviation helmet is considered a special type of headband.

volume of the enclosed space, headband pressure, and cushion leaks. The earphone socket serves the dual role of coupling the earphone to the ear and of excluding unwanted noise. Here again, the socket's performance is modified by secondary factors, such as the size and weight of the earphone and the headband pressure. From such considerations it is clear that the functioning of each component depends in part upon the combination that is used.

The headset which provides a satisfactory speech-to-noise ratio must have good insulation and uniform frequency response. The headsets with which we began World War II were inferior on both counts. The Army Air Forces

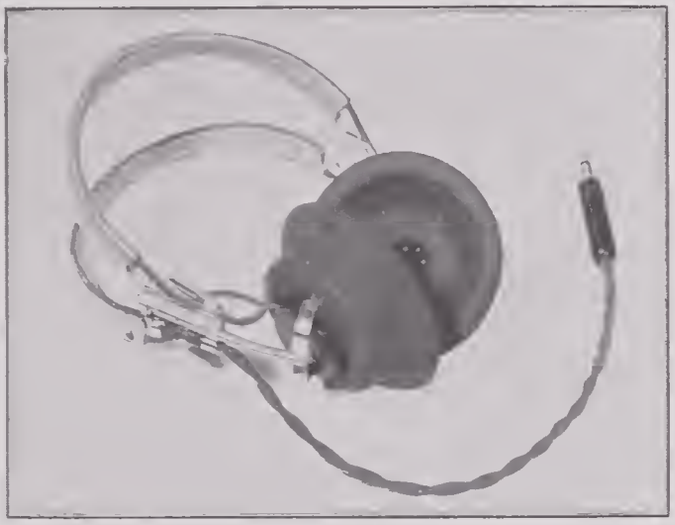


FIGURE 28. Headset HS-23.

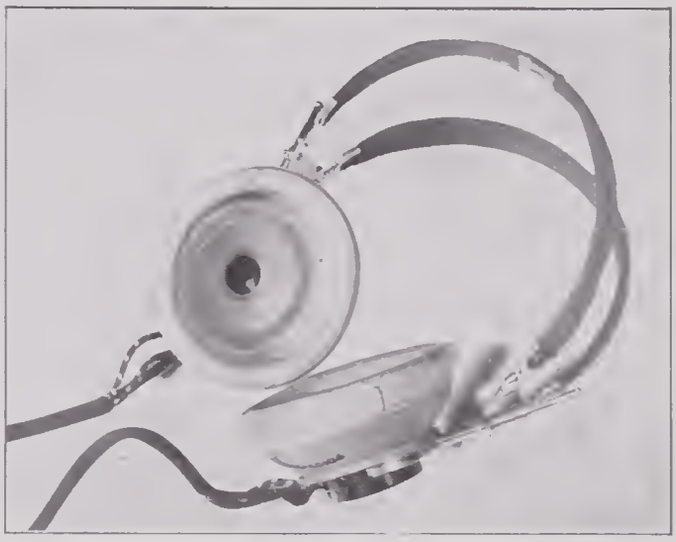


FIGURE 29. Headset NAF-38610.

headset, HS-23 (see Figure 28), combined highly resonant earphones, R-14, with comfortable, sponge rubber sockets which provided little insulation against frequencies below 1,000 c. The Navy headset, NAF-38610 (see Figure 29), also used resonant earphones, CTE-49015 (TH-37), with poorly insulating sockets, CMO-49104 (TC-66), and a headband which did not supply sufficient pressure to seat the sockets properly.

A series of studies on the necessary characteristics for headsets,[2] together with the active cooperation of leading manufacturers, led to the development of magnetic and dynamic earphones with uniform response up to 4,000 c. Earphone sockets designed by the Bell Telephone Laboratories and by the Psycho-Acoustic Laboratory provided greater insulation at all frequencies. Such improvements made it possible for the AAF to adopt the magnetic earphone, ANB-H-1, and the Bell Laboratories' socket, MX-41/AR, as components in their later headset HS-33 (see Figure 30). Although these sockets provided an excellent seal, they were not so comfortable as the Harvard Socket, M-301. The Harvard design, a circumaural type of "doughnut" socket, was optional for headset HS-33 but was not generally issued because it was feared that the lower sensitivity of the headset could not be made up in additional power. Furthermore, the chamois cushion did not withstand heat, dirt, and humidity well, a consideration in tropical and ground force applications. The Harvard design was adopted, however, for use in aircraft helmets.

The Navy adopted a dynamic earphone, ANB-H-1A, designed to meet the same specifications as the Army's magnetic ANB-H-1. This dynamic earphone, when combined with an improved headband and cushions of the doughnut type, made up the H-4/AR headset (see Figure 31).

Early in the war the Army Ground Forces adopted headset HS-30, shown in Figure 32. The response of the earphones R-30 fell off badly above 2,500 c, and the headband and tip were not entirely satisfactory. When properly adjusted, however, the semi-insert tip furnished good protection against ambient noise. The

semi-insert tip was retained, somewhat less effectively, in the later headset H-16/U (see Figure 33). The semi-insert coupler in this case is used in addition to a circumaural socket. The poor frequency response of the R-30 unit was retained, but the headband was greatly improved. The H-16/U has by far the most complicated construction of any of the headsets considered herein. Altogether, the poor frequency response of the R-30, the serious discomfort of the HS-30, and the precarious seal provided by the H-16/U make these headsets the least satisfactory of any types now in use.

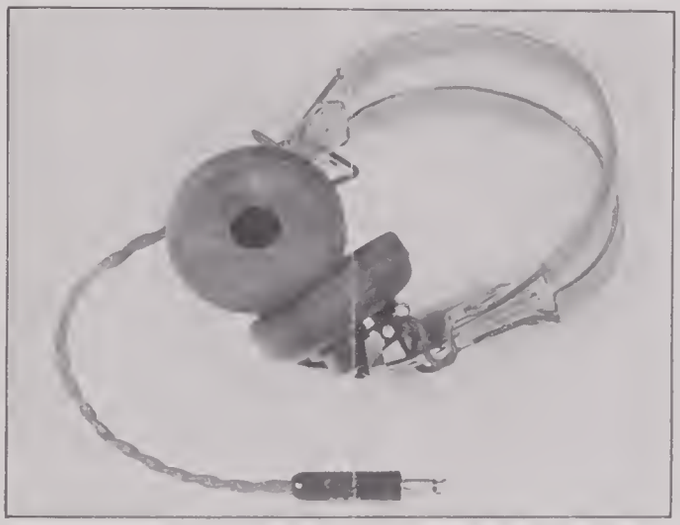


FIGURE 30. Headset HS-33.

FIGURE 31. Headset H-4/AR.

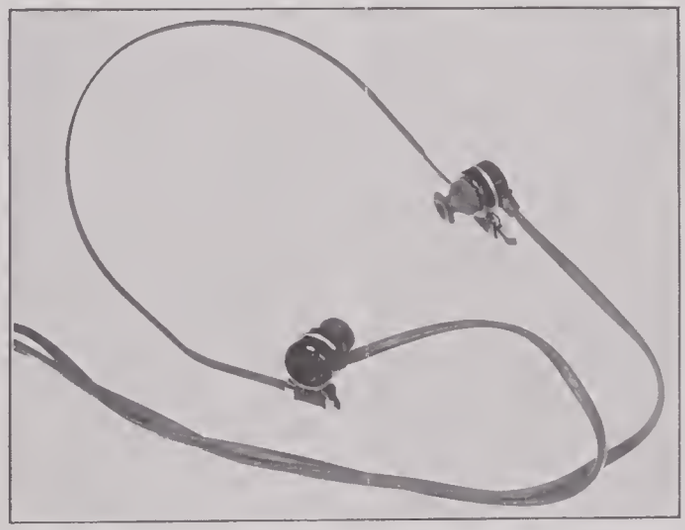


FIGURE 32. Headset HS-30.

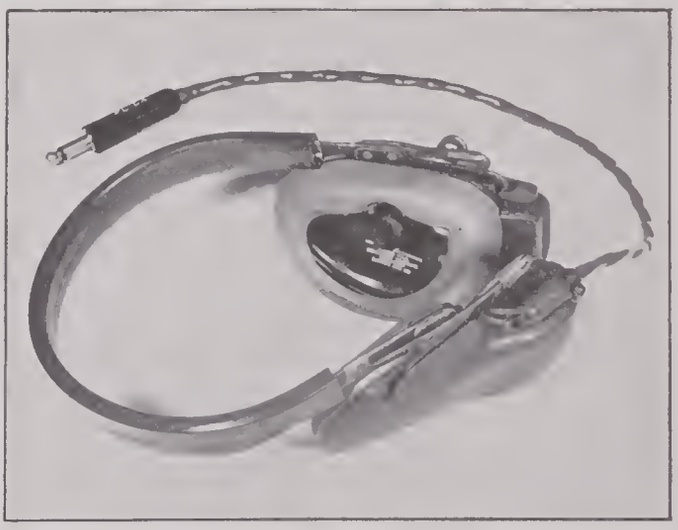


FIGURE 33. Headset H-16/U.

In general, circumaural sockets are more comfortable, but they provide a larger volume and less efficient coupling between earphone and eardrum than supra-aural cushions. The volume driven by an earphone in the supra-aural MX-41/AR socket is about 5 to 6 cu cm; with the Harvard circumaural sockets it is about 22 cu cm. The effect of the larger volume is essentially a loss in sensitivity of 5 to 10 db, depending upon the type of earphone. A more recent circumaural socket of Harvard design fits closely around the ear and thus provides the advantage of comfort along with a relatively small enclosed volume (about 11 cu cm). This socket was adopted by Canada and is shown in Figure 34.

Comfort is likewise an important factor in determining headband pressure. Lowering the pressure increases the comfort, but decreases the acoustic efficiency of the headset by allowing leakage around the socket. Leaks permit ambient noise to reach the listener's ears, and also produce a low-frequency resonance between 100 and 1,000 c. Data are available[12] which indicate the pressure necessary for each type of socket.

Articulation tests with dynamic (ANB-H-1A) and magnetic (ANB-H-1) earphones in

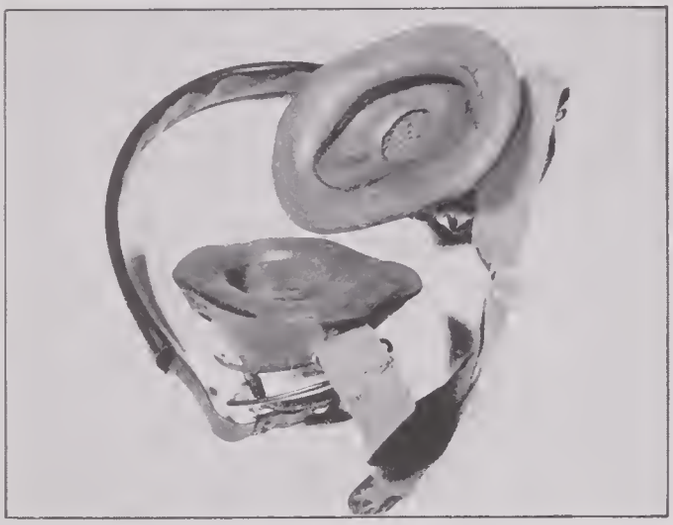


FIGURE 34. Canadian headset ZA/CAN 1637 with Harvard design 8-C earphone socket.

three types of sockets[3] showed that the decreased sensitivity of sockets enclosing a large volume is directly reflected in the articulation scores. If the volume is large, more gain is needed in the interphone amplifier. The same tests also demonstrated that the dynamic earphones are slightly superior to the magnetic, regardless of which earphone sockets are used.

9.5 THE INTERPHONE AT ALTITUDE

As our fighting planes went to higher and higher altitudes, new problems and new complications were added to the already difficult task of communication in high ambient noise. Frequency-response characteristics change, sensitivity decreases, and talkers find it arduous or impossible to maintain an adequate voice level. Failure of interphone communication at altitude was a regular occurrence in the early days of high-altitude flying.

The various ills which afflict the interphone at altitude can be conveniently summarized in the term *altitude decrement* [AD]. The altitude decrement of an interphone is the sum of the decrements of its components, including the effects exhibited by listener and talker. In one way or another the altitude decrement can be traced back to the effects of reduced ambient pressure. Temperature changes from 20 C at 5,000 ft to — 20 C at 35,000 ft are also encountered,[8a] but tests have shown that changes due to temperature are of less importance to interphone performance than are pressure changes. Altitude decrement can thus be defined as the change in performance due to the reduction of the ambient pressure.

The effect of altitude on the human voice was described in Section 4.1.3 for the case where the talker does not wear an oxygen mask. Figure 14 in Chapter 4 shows the average altitude decrement for the three vowels ū, a, and ē, which is about 10 db between 400 and 4,000 c.

A more practical problem, however, is to determine the talker's behavior when he is wearing an oxygen mask. Experiments were undertaken by the Electro-Acoustic Laboratory[4] with a crew of trained men in an altitude chamber. These talkers spoke with "half effort" (see Section 4.1.3) at sea level and at altitude, and an audio-spectrometer was used to analyze the speech sounds. Several types of oxygen masks were used with a magnetic mask microphone (MC-253-A). The magnetic instrument avoids the complication of nonlinearity inherent in carbon microphones. The altitude decrement for the microphone (see Section 10.5) was then subtracted from the total decrement of voice-mask-microphone to obtain the decrement of the voice-mask combination.

The results of this investigation showed that the altitude decrement of the voice-mask combination is largely independent of the type of oxygen mask. The decrement is different, however, for different speech sounds. As a general rule the average altitude decrement of a given voice-mask combination at altitude

A_2 relative to altitude A_1 is given approximately by the expression,

$$\text{AD} = 10 \log \frac{\text{air density at } A_1}{\text{air density at } A_2}.$$

Figure 35 shows the average altitude decrement at 35,000 ft with a number of talkers

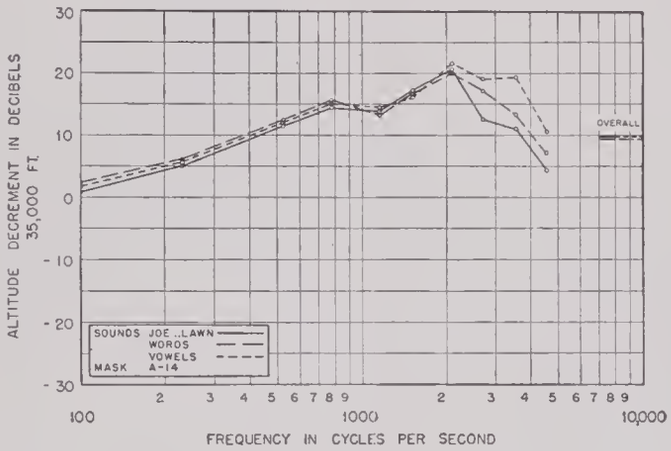


FIGURE 35. Average altitude decrement, experienced by talkers using oxygen masks, for various speech sounds at 35,000 ft. (The decrement of the microphone is not included.)

for different speech materials. Comparison with a similar curve obtained without an oxygen mask (see Figure 14, Chapter 4) reveals that the effect of the mask is to boost the levels at low frequencies. Between 400 and 4,000 c,

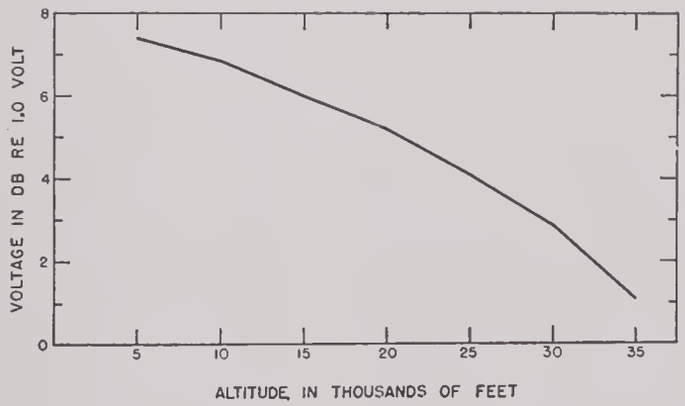


FIGURE 36. Peak instantaneous voltage at microphone terminals during articulation tests conducted in flight. (ANB-M-C1 microphone in A-14 oxygen mask.)

however, the presence of the mask increases the altitude decrement.

It should be stressed that these data were obtained with the talkers maintaining the same subjective effort at altitude as at sea level. In actual practice the talker monitors his voice in his own earphones, and when his voice level drops at high altitudes he will exert additional effort to maintain an adequate level. Thus his altitude decrement will be somewhat smaller than indicated above. For engineering purposes, the "half-effort" decrement can be regarded as including a desirable safety factor.

The decrements which are actually observed[8b] under flight conditions, where the talker monitors his voice level by his own side-tone, are indicated in Figure 36. These measurements, made in a B-17 bomber at Eglin Field, represent the decrement of the voice-mask-microphone combination as a function of altitude. Eleven announcers with ANB-M-C1 microphones in A-14 oxygen masks were used. The altitude decrement in this case amounts to about 6 db at 35,000 ft.

The variability in speech level from word to word for a given talker is approximately the same at altitude as it is at sea level. Differences between talkers are slightly greater at altitude than at sea level. The range of signal levels which the interphone amplifier is required to handle is, therefore, about the same at all altitudes; it is only the mean speech level which changes.

The altitude decrement of the voice-mask-microphone combination does not represent the total decrement for the interphone. The sensitivity of the earphones also changes at high altitudes. As the atmospheric pressure is reduced, the compliance of the air volumes increases, but the impedance of the diaphragm remains the same. The effect of this change depends upon the design of the particular earphone considered, but in general the frequency response is altered and the sensitivity is decreased (see Section 10.5).

In order to evaluate the effect of these combined decrements upon articulation efficiency under actual conditions of flight, an extensive testing program was undertaken at Eglin Field, Florida,[5, 6, 8] with the cooperation of the Aircraft Radio Field Laboratory and of the Army Air Forces Proving Ground Command. A representative testing crew of AAF men was selected and trained. During flights in B-17F bombers, the test crew was stationed as indicated in

Figure 37. Two experimenters participated in all the flights.

Typical results from articulation tests conducted under these conditions are presented in Figure 38.[h] The data were obtained with the ANB-H-1 earphones and an experimental interphone amplifier which permitted adjustment of the voltage gain in the system. In Figure 38

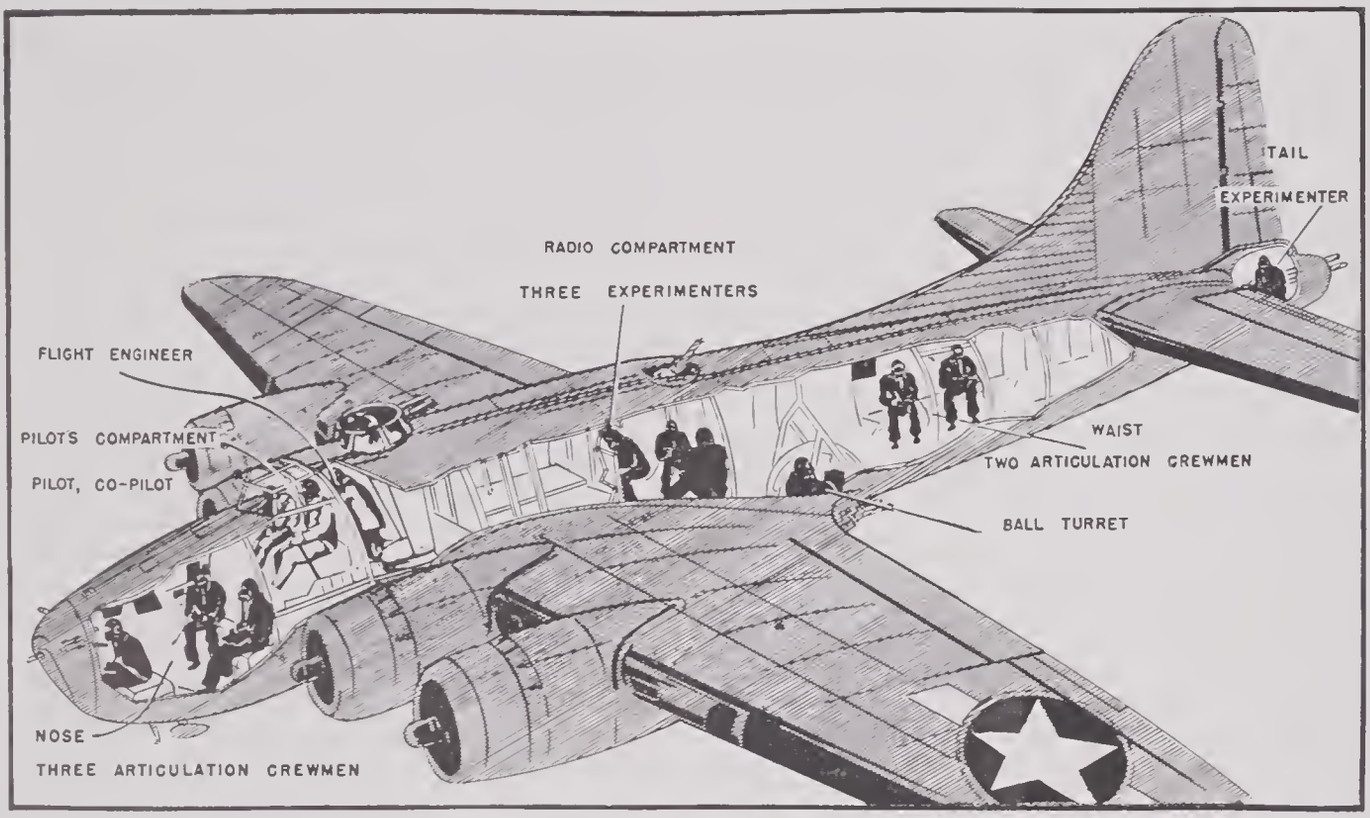


FIGURE 37. Crew positions in B-17F.

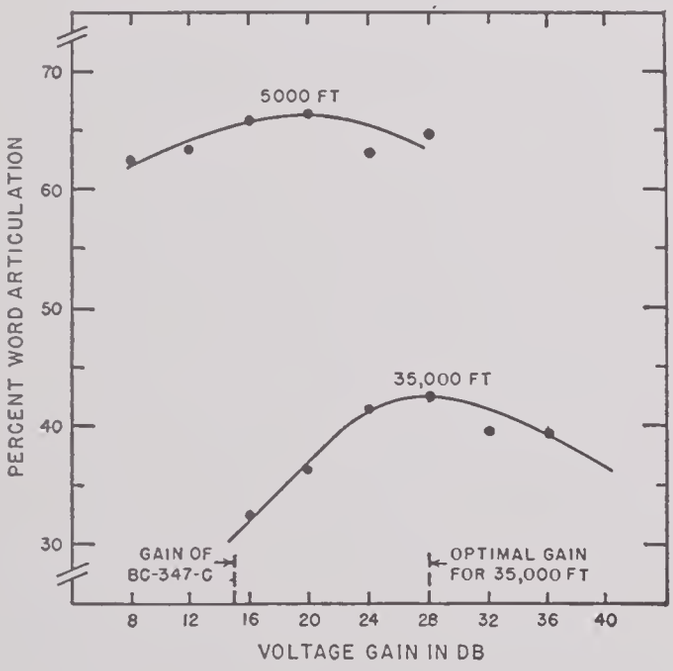


FIGURE 38. Functions showing the relation between word articulation and voltage gain at 5,000 and 35,000 ft.

the articulation of the system is plotted as a function of the voltage gain of the interphone amplifier, with altitude as the parameter. From these functions it is apparent that, by boosting the amplifier gain at high altitude, about one-third to one-half of the altitude decrement can be recovered.

On the basis of these and similar tests conducted both at Eglin Field and at the Psycho-Acoustic Laboratory, recommendations were made for interphone amplifiers (see Section 9.3) to give the optimal performance at altitudes. It is obvious, of course, that the advantage from increasing the voltage gain of the amplifier is counteracted by overload distortion if the power output is limited. The tests showed

[h] The difference in performance between 5,000 and 35,000 ft shown in Figure 38 is somewhat larger than the difference indicated for comparable equipment in Figure 1 of this chapter. The larger difference in Figure 38 resulted from the use of more difficult test words in this series of tests.

that the peak power per head set should be at least 200 milliwatts, preferably 400 milliwatts. The improvements in articulation which result from increased gain and power constitute very real advantages to bomber personnel to whom the interphone represents the key to teamwork necessary for the successful completion of a mission.

Chapter 10

TEST METHODS AND EQUIPMENT FOR INTERPHONE COMPONENTS

IN THIS CHAPTER a list of test methods and equipment which are applicable to interphones and their components is given. This list covers all the types of tests which were found useful, with the exception of the articulation test discussed in Chapter 5.

A considerable part of the activities of the Psycho-Acoustic and Electro-Acoustic Laboratories during World War II was spent in assisting the Services (represented chiefly by the Army Air Forces and the Navy, Bureau of Aeronautics) in the improvement of aircraft communications as a whole, and in the procurement of suitable instruments. To accomplish this, the Harvard group functioned as an independent test laboratory which, at the request of the Services, assisted manufacturers of interphone equipment in the specific problems they had to meet. The results of various tests are contained in an extensive series of reports.[9, 11, 13-15, 17, 18, 20-24, 25, 27, 28, 30]

10.1 METHODS AND EQUIPMENT FOR TESTING MICROPHONES USING HUMAN TALKERS

The microphones in use in military communications are, without exception, closely coupled acoustically to the talker, and the resulting interaction causes the performance of the microphone to vary according to the talker using it. For reasons of simplicity and speed, therefore, an artificial voice is often substituted for the talker in laboratory tests (see Section 10.2).

In many cases the substitution in laboratory tests of an artificial voice for the talker is permissible, but in other instances (throat microphones, oxygen masks, noise shields, etc.) such a procedure is questionable and subject to large errors. Consequently, it is sometimes necessary to use the more elaborate procedures involving the use of real talkers wearing or using the equipment.

10.1.1 Frequency Response

The real-voice response (see Section 9.1.3) of a microphone can be considered a valid measure of the response to speech under representative conditions of use. Extensive tests of a variety of types of microphones were carried out at the Electro-Acoustic Laboratory, using a crew of three talkers and the audio-spectrometer (see Section 4.1.1). The standard test sentence, "Joe took father's shoebench out; she was waiting at my lawn," devised by the Bell Telephone Laboratories, was used.[38]

The talkers were seated in a room free from acoustic wall reflections. The free-field sound pressure at a distance of 1 m in front of the talkers' lips was measured by means of a condenser microphone (WE Type 640-AA) and sound-pressure meter (see Section 10.6). The output was analyzed by an audio-spectrometer in 12 contiguous frequency bands covering the range from 60 to 9,000 c. The long-time average value of the square of the microphone output voltage in each frequency band was measured simultaneously in the 12 channels. A thirteenth channel measured the overall value.

The talker was then instructed to use the microphone under test. Its output voltage was analyzed by the audio-spectrometer. The ratio, in decibels, of the microphone output voltage in a band to the corresponding free-field sound pressure yields the real-voice response of the microphone (see Section 7.1.3). The talker was instructed to talk with what he judged was "constant effort" with and without the microphone under test. Certain vowels were used for this purpose. A standard volume indicator was used to adjust the overall average level of speech to a free-field sound-pressure level of about 74 db when talking into open air.

If the response of a microphone so obtained is to be a valid measure of performance, certain precautions in experimental technique and evaluation of the results are necessary. For example, the results obtained with carbon in-

struments have to be interpreted with caution,[a] since the results may depend on the level of speech, the spectrum of the speech signal, position of the instrument with respect to gravity, carbon current, and history of the carbon granules. Breath noises must be guarded against. The type of speech sounds used to test the microphone is important in all cases. Special attention must be given to this problem in the case of throat microphones where response to consonants may easily be simulated by response to harmonics of strong vowel sounds. For a more valid test, special speech material may have to be used.

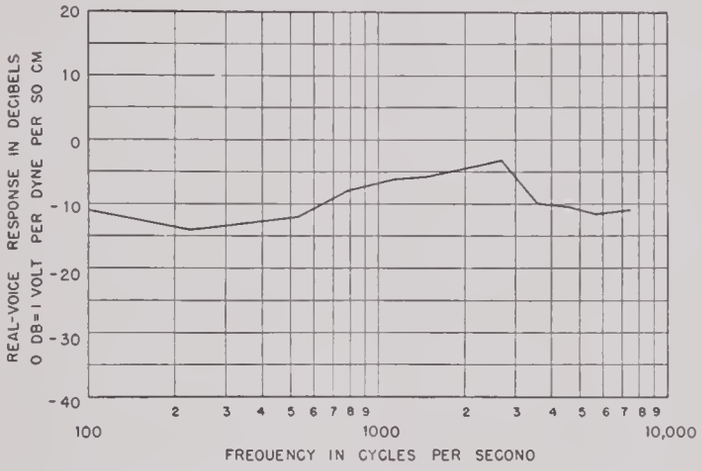


FIGURE 1. Real-voice response of a type T-17 carbon microphone.

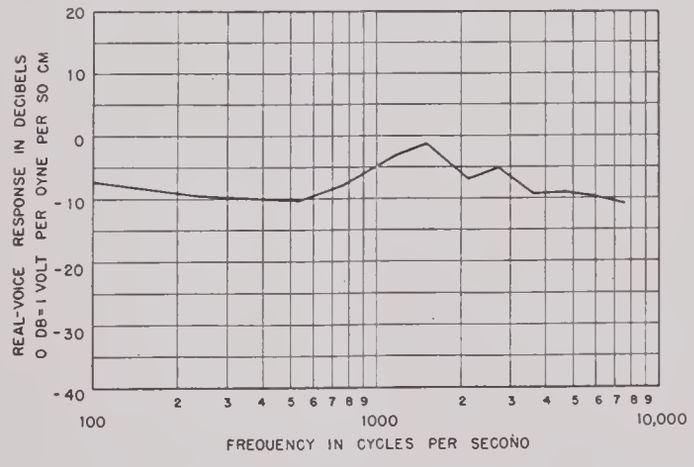


FIGURE 2. Real-voice response of a Type T-38 carbon microphone.

Detailed data are available for about 40 microphones tested at the Electro-Acoustic Laboratory.[38] Figures 1 through 8 show the real-voice responses for several microphones. They include microphones mounted in a noise shield (Harvard Design D-18) and in an oxygen mask (A-14). Attention should be

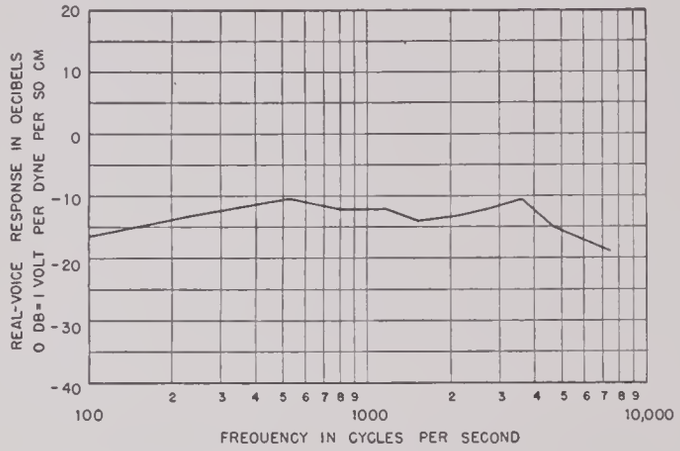


FIGURE 3. Real-voice response of a type M-6/UR carbon noise-canceling microphone.

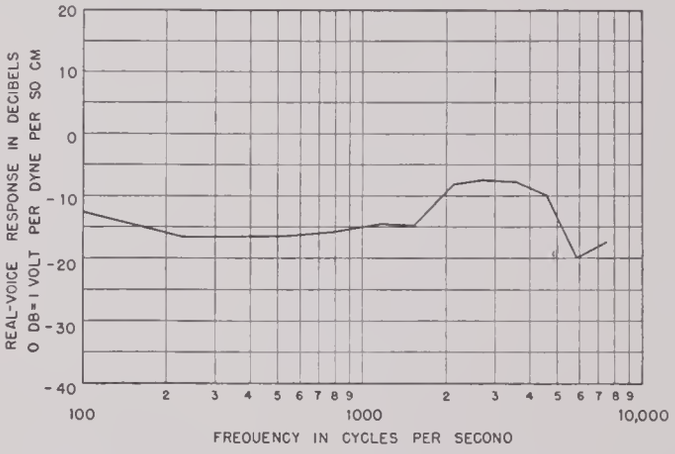


FIGURE 4. Real-voice response of a type T-45 carbon noise-canceling microphone.

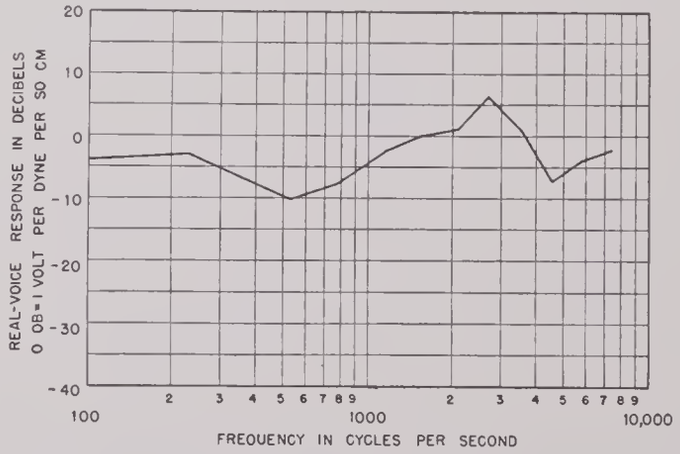


FIGURE 5. Real-voice response of a type ANB-M-C1 carbon microphone in a Harvard type D-18 noise shield.

[a] This is true with real or artificial voices.

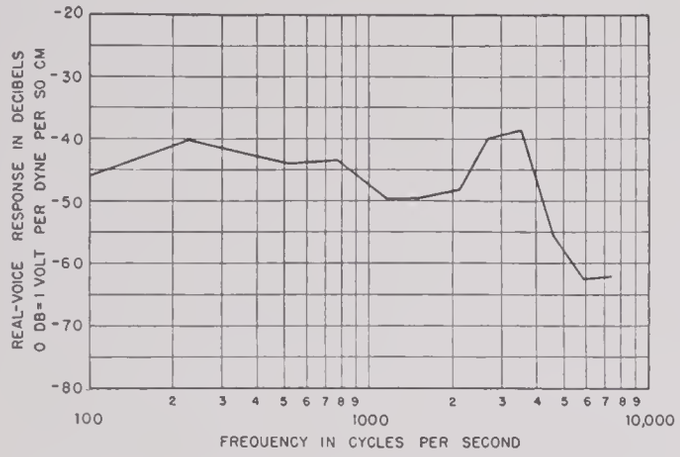


FIGURE 6. Real-voice response of a type MC-253A magnetic microphone in a Harvard type D-18 noise shield.

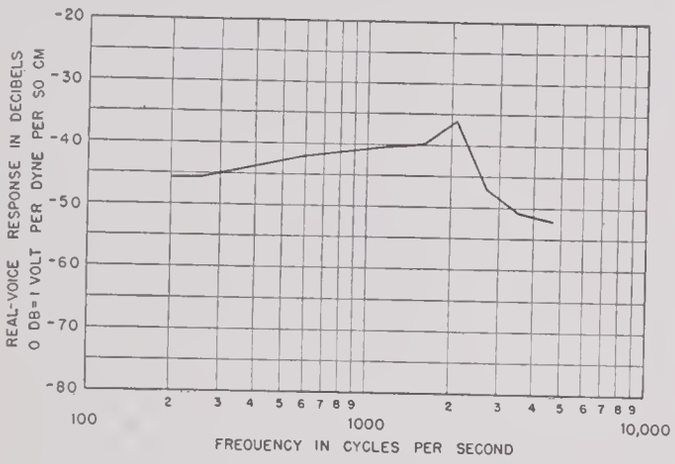


FIGURE 7. Real-voice response of a type MC-253 magnetic microphone in a type A-14 oxygen mask (sea level).

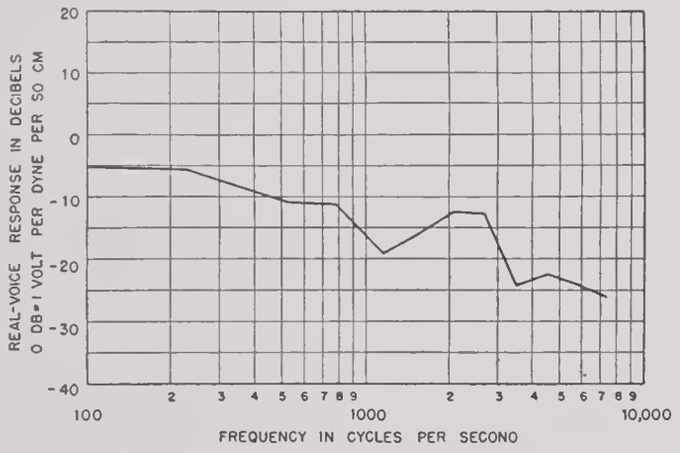


FIGURE 8. Real-voice response of a type T-30-V carbon throat microphone.

drawn to the fact that the response of the microphone element used in such enclosures must suitably complement the acoustic effects of the enclosure for acceptable performance as a combination. This entails a drop in microphone output for frequencies below about 2,000 c at a rate of about 6 db per octave when the microphone is measured as a separate unit.

10.1.2 Insulation against Ambient Noise

Tests on a number of microphones were performed at the Electro-Acoustic Laboratory to evaluate the noise picked up in a random sound field.[38a] The talker (who was silent during the noise measurements) was seated in a room equipped with hard polycylindrical protrusions

FIGURE 9. Room with polycylindrical wall configurations for tests in random noise fields.

(Figure 9). A random noise field of 85-db spectrum level (approximately uniform with frequency) was set up, and the output voltage of the microphone due to noise was measured with an audio-spectrometer (see Section 9.1.4). For carbon instruments, the procedure was repeated for a spectrum level of 70 db. Figures 10 through 15 show the output voltage per cycle of six microphones in noise.

Clearly, the speech-to-noise ratio of a microphone can be determined directly for a given

speech level and noise spectrum by analyzing the output voltages due to speech and noise. By suitable choice of test conditions, meaningful comparisons between microphones are easily made (see Section 9.2). The exact characteristics of the analyzing device are of secondary importance, since only voltage ratios are measured. Audio-spectrometers with both

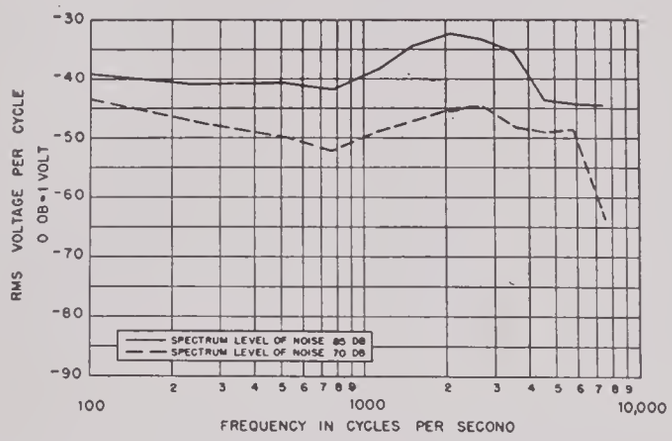


FIGURE 10. Voltage generated by a type T-17 carbon microphone used in a random noise field of uniform spectrum level.

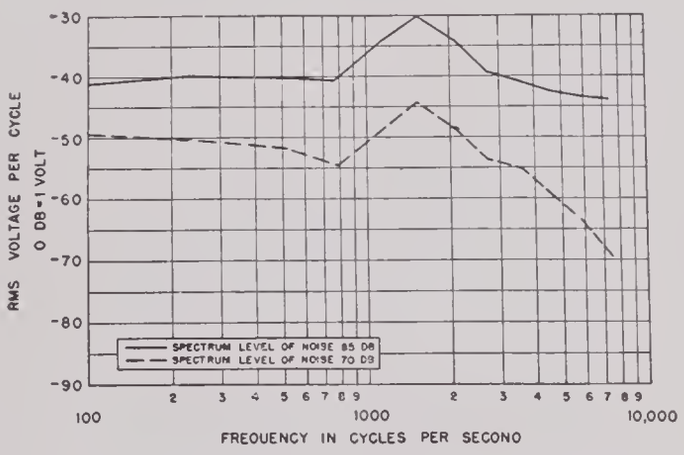


FIGURE 11. Voltage generated by a type T-38 carbon microphone used in a random noise field of uniform spectrum level.

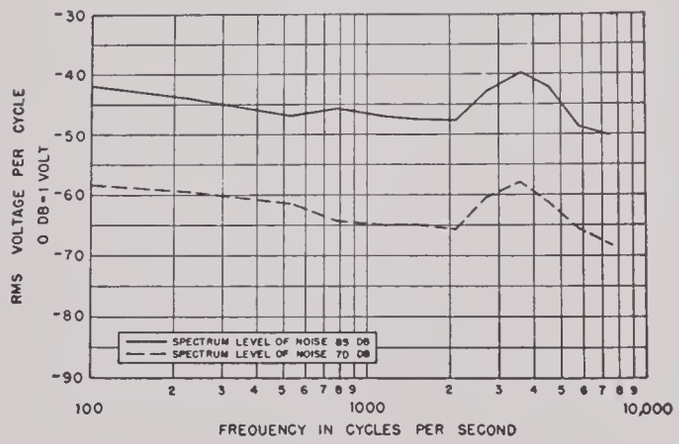


FIGURE 12. Voltage generated by a type M-6/UR carbon noise-canceling microphone used in a random noise field of uniform spectrum level.

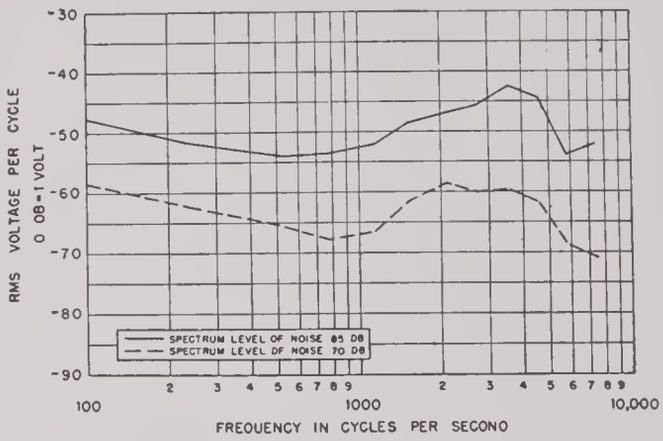


FIGURE 13. Voltage generated by a type T-45 carbon noise-canceling microphone used in a random noise field of uniform spectrum level.

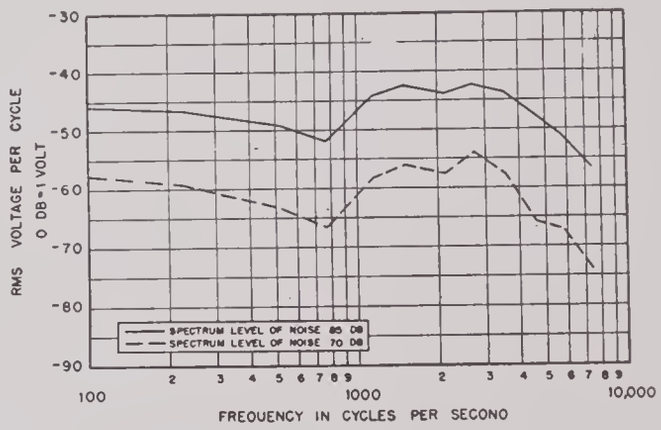


FIGURE 14. Voltage generated by a type ANB-M-C1 carbon microphone in a Harvard type D-18 noise shield used in a random noise field of uniform spectrum level.

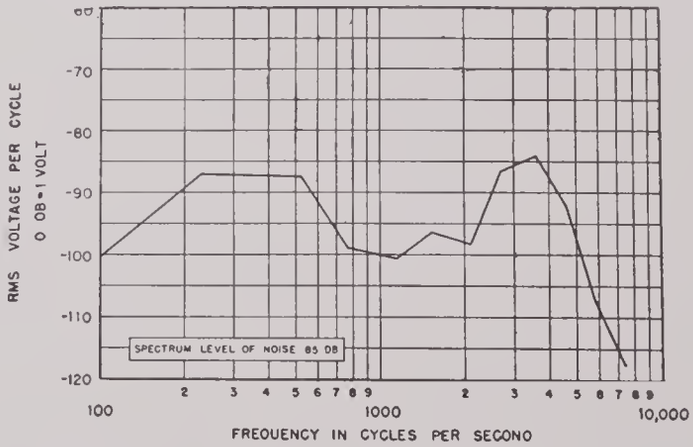


FIGURE 15. Voltage generated by a type MC-253A magnetic microphone in a Harvard type D-18 noise shield used in a random noise field of uniform spectrum level.

linear and square-law characteristics have been used successfully for such measurements.

It has been observed that if a microphone is very closely coupled to the mouth the speech-to-noise ratio is no longer independent of the movements and position of the mouth of the talker for high levels of ambient noise. The noise-canceling lip microphone (Type M-6/UR) provides a good example of this effect. A high-speed-level recorder was used to record the microphone voltage due to speech and the voltage due to noise as modified by the configuration of the talker's mouth while he silently articulated the test sentence, "Joe took father's shoebench out; she was waiting at my lawn," with long pauses between the words.[34] It was found that the speech-to-noise ratio so obtained was, on the average, about 5 db lower than the "static" value in which the talker kept his mouth closed throughout the noise level measurement.

These tests have a direct bearing on the use of voice-operated relays which have been suggested for use in very high ambient noise levels. Such relays would protect the interphone and crew from the constant noise picked up between messages. It has been found,[33] however, that the speech-to-noise ratio provided by the M-6/UR in a noise level prevailing in a F7F Navy airplane (see Chapter 2) is not sufficient for reliable operation of such a relay. Some measure of success was had by the substitution of an ANB-M-C1 microphone in an A-14 oxygen mask, *provided* care was taken to obtain a perfect seal of the mask around the face and consequently more favorable noise exclusion.

For tests of the type described above, and also for articulation tests in ambient noise, suitable electric noise generators have been developed at the Psycho-Acoustic and Electro-Acoustic Laboratories. Random noise is conveniently produced by gaseous discharge tubes[10] or by using the thermal noise across the terminals of a resistor.[38b] To simulate the low-frequency components present in the spectrum of noise in airplanes and other combat vehicles, sharp pulses are injected into the random noise. The generator producing these pulses is usually synchronized with the power-line frequency of 60 c.

10.2 METHODS AND EQUIPMENT FOR TESTING MICROPHONES USING AN ARTIFICIAL VOICE

For testing a large number of microphones in a production testing laboratory, for quickly observing the effect of a modification in the design, and for ready comparison of a number of similar units, simplified test methods which eliminate human talkers are required. The talker is replaced by a suitably designed sound source, or artificial voice, and the response of the microphone under test is expressed in terms of the output voltage developed for a given sound-pressure level. In some procedures, the sound pressure is specified as the free-field sound-pressure level 1 m directly in front of the voice. In others, the sound-pressure level at the microphone face (grid) is specified. The advantage of the methods using an artificial voice lies primarily in simplicity, speed, and versatility at the expense of validity. Clearly,

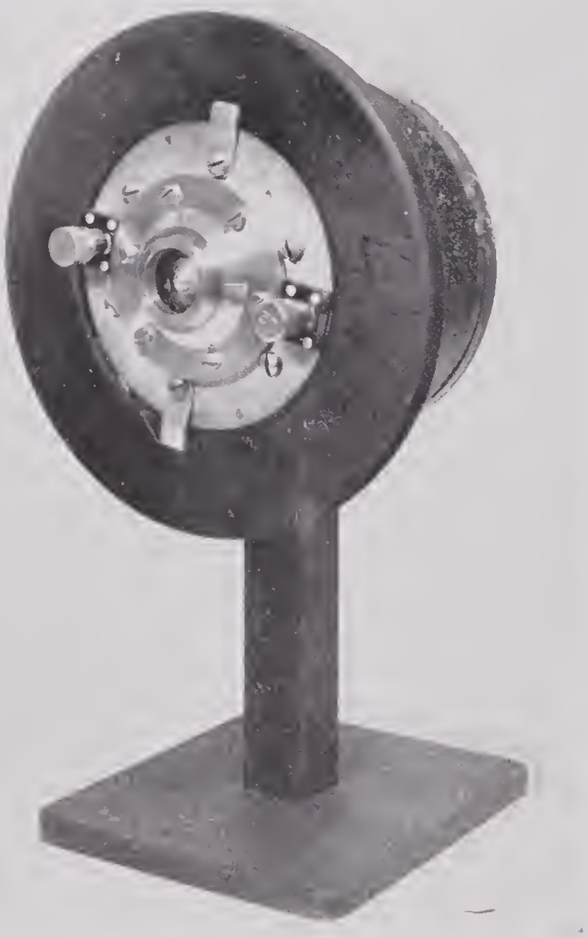


FIGURE 16. Photograph of the artificial voice standardized by the Joint Radio Board [JRB].

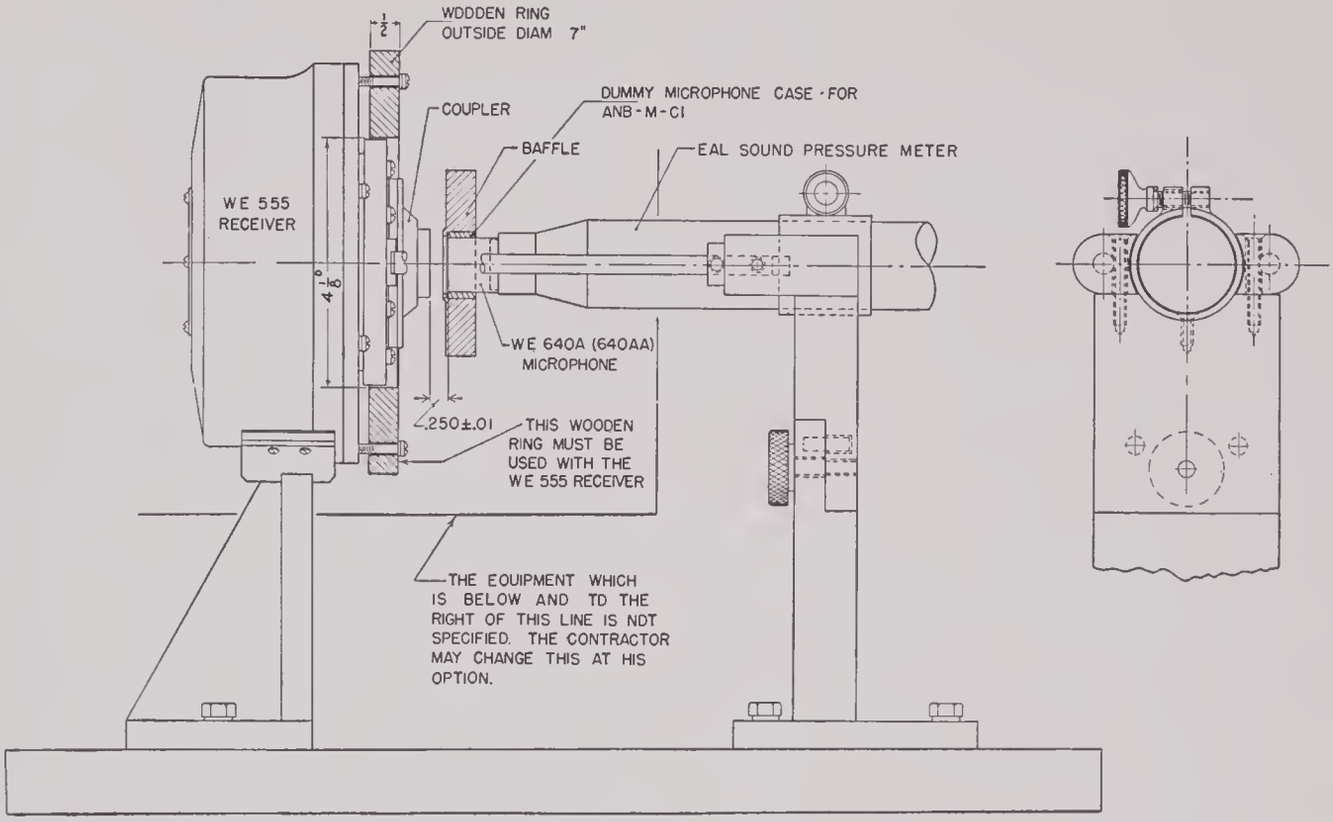


FIGURE 17. Drawing of the artificial voice (JRB) showing the microphone in position for calibration.

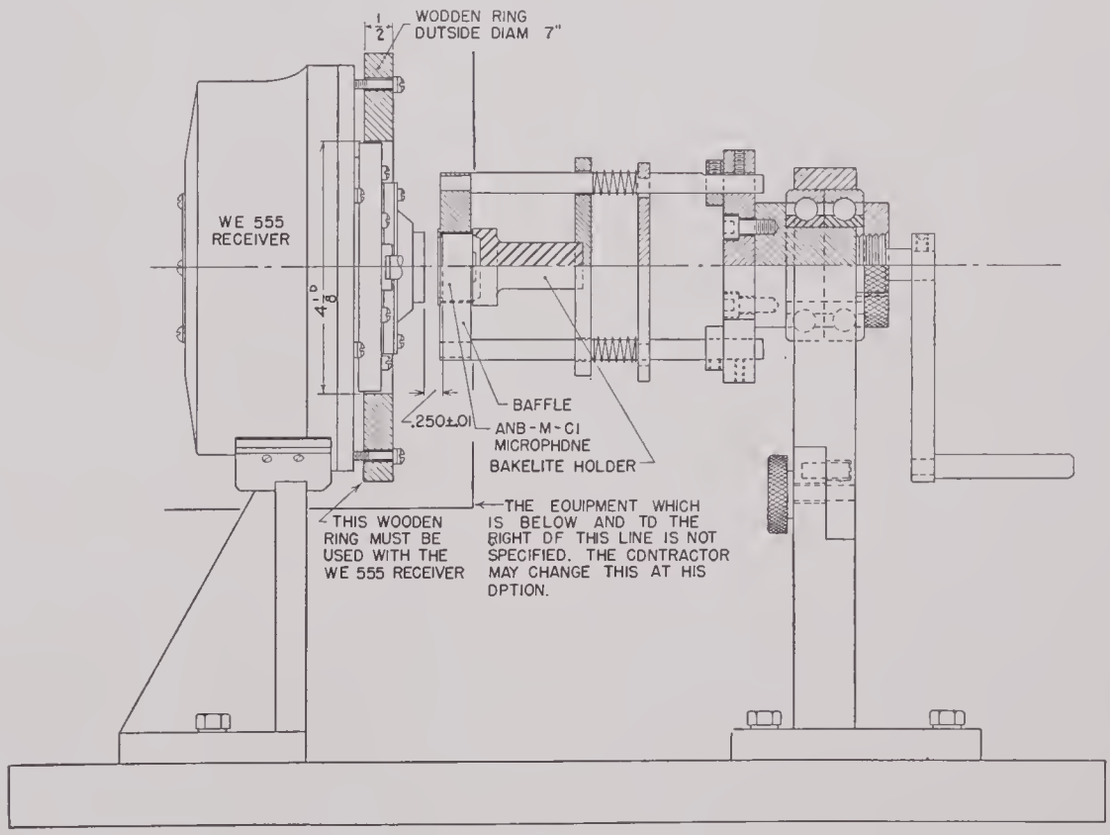


FIGURE 18. Drawing of the artificial voice (JRB) showing a type ANB-M-C1 microphone in position for response measurements.

throat microphones cannot be tested by this method. The results obtained with microphones mounted in an enclosure or with instruments whose performance depends critically upon the shape and configuration of the talker's mouth and head are, as a rule, not valid in terms of real-voice responses. Nevertheless, even in those cases, measurements with artificial voices do provide the designer with a laboratory tool of some merit.

10.2.1 Frequency Response

The Joint Radio Board of the Joint Aircraft Committee [JRB] standardized in 1943 the test procedure and equipment for measuring the frequency-response characteristic of microphones. This action was based largely on recommendations submitted by the Electro-Acoustic Laboratory. The procedure applies to microphones used in open air.

The artificial voice (see Figure 16) consists of a modified dynamic loudspeaker (driving unit WE 555) and is driven by a suitable power amplifier and audio-frequency oscillator. To calibrate it, a baffle 3 in. in diameter is used to set up a constant sound-pressure level of 115 db at its center by means of a condenser microphone (WE 640-A or 640-AA) and a sound-pressure meter (see Section 10.6). To simulate the microphone under test, a dummy microphone case is used. The distance of the baffle from the edge of the artificial voice is ¼ in. The apparatus with baffle and sound-pressure meter in place is shown in Figure 17. After calibration of the artificial voice the dummy microphone case with the condenser microphone is removed and the microphone under test is substituted. The output voltage of the unknown microphone is then determined as a function of frequency (see Figure 18).

For carbon instruments, suitable conditioning and agitating procedures are prescribed. They are designed to provide a uniform condition of the carbon granules, and to provide also for a simulated agitation similar to the one due to the ambient noise in conditions of actual use.

Engineering drawings and full details of the testing procedure are to be found in reference 19a.

Test apparatus of a somewhat more elaborate form, which permits testing of carbon instruments at various positions, is shown in Figure 19, with a set of baffles and dummy cases.

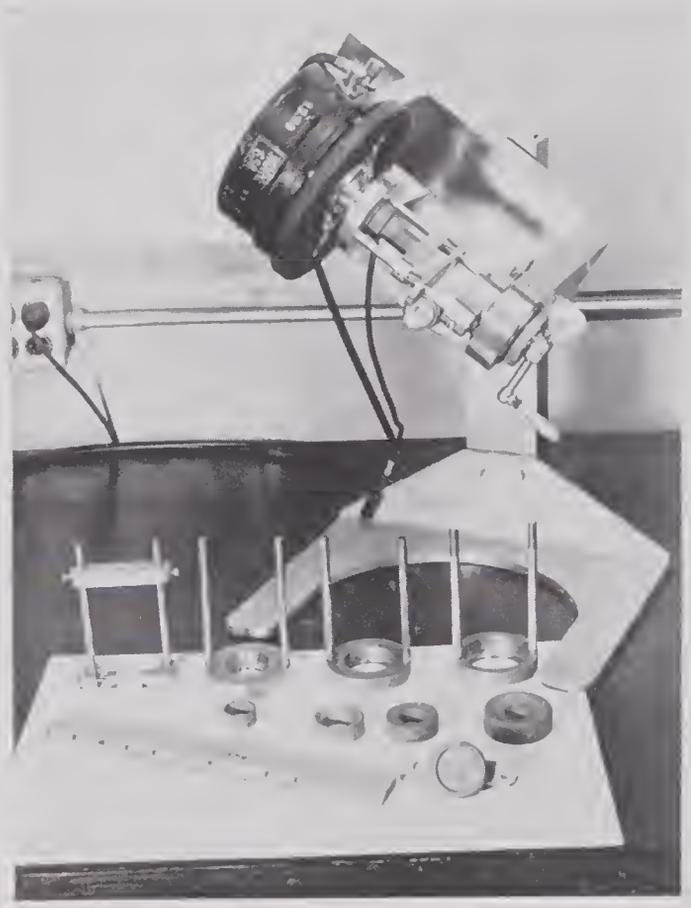


FIGURE 19. Artificial voice with various baffles and dummy microphone cases, suitable for determinations of the position effect in carbon instruments.

To facilitate comparison with real-voice response characteristics, the results obtained by the JRB technique were expressed in terms of voltage developed per unit free-field sound pressure at a distance of 1 m from the artificial voice. This was roughly achieved by using the inverse square law to convert from a distance of ¼ in. to 1 m. Clearly, this procedure is only very approximate; besides, it ignores the nonlinearity of carbon instruments and the obstacle effects of the microphone housing.

In Figures 20 through 23 a comparison is given between the real-voice response and the artificial-voice response of a T-17, T-38,

M-6/UR, and T-45 microphone. Differences of this type and magnitude justify, in many cases, the use of an artificial voice. Note that these microphones are designed for operation in open air.

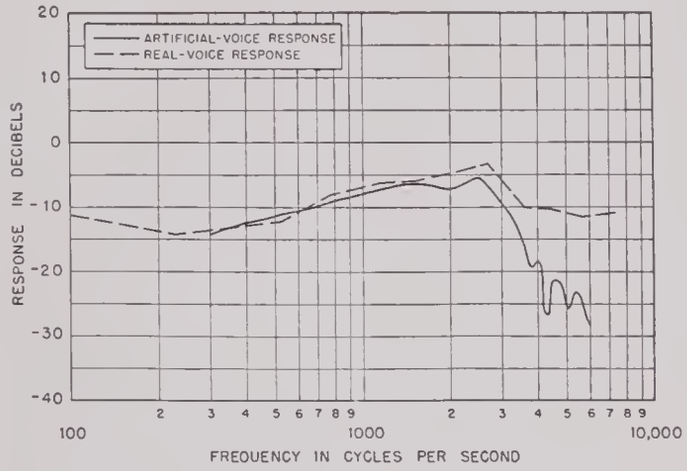


FIGURE 20. Showing a comparison between real-voice and artificial-voice response of a type T-17 carbon microphone.

Figures 24 and 25 show artificial-voice calibrations of ANB-M-C1 and MC-253 microphones tested in open air outside the enclosures for which they are designed. Note the rising characteristic at low and medium frequencies.

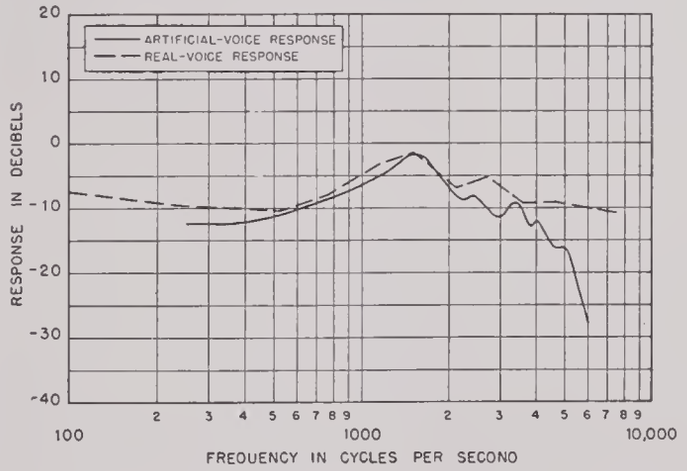


FIGURE 21. Showing a comparison between real-voice and artificial-voice response of a type T-38 carbon microphone.

This rise compensates for the enhanced response in that range when the microphones are used in an enclosure. Figures 26 and 27 show response characteristics of two carbon instruments taken at various levels of input sound-pressure level. Response curves in a large number of instruments are given in report form.[19b]

Clearly, the artificial voice described is not suitable for testing oxygen masks and noise shields. For this purpose, an artificial voice was designed at the Electro-Acoustic Laboratory.[19c] It consists principally of a loudspeaker driving unit (WE 555) and a dummy head (see Figure

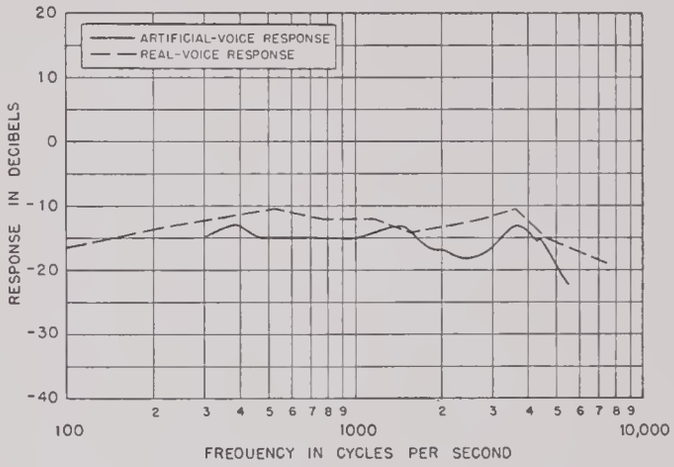


FIGURE 22. Showing a comparison between real-voice and artificial-voice response of a type M-6/UR carbon noise-canceling microphone.

28). A tube of high acoustic impedance is brought through an opening in the head and terminated in the mouth of the dummy. The tube diameter (¾ in.) is comparable with the dimensions of the opening in the human mouth

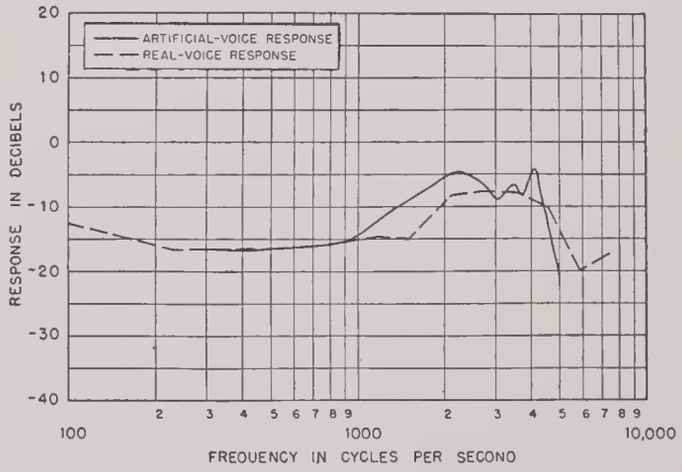


FIGURE 23. Showing a comparison between real-voice and artificial-voice response of a type T-45 carbon noise-canceling microphone.

during speech. To obtain a high internal impedance, the brass tube is partially filled with an absorbing material (Fiberglas). The dummy head is made of plaster and covered

with about ½ in. of Korogel. The shape of the head was one of a series designed for the Aero-Medical Laboratory at Wright Field, Dayton, Ohio, and represents the average measurements of a large number of individuals.

Because of the high impedance of the artificial voice, it is not feasible to generate signal levels approximating those found in use (115 to 120 db inside the mask). A magnetic mask microphone (MC-253-A) was used in the measurements instead of the usual carbon instrument (ANB-M-C1), since the linearity of the magnetic unit permitted extrapolation to the high signal levels. An artificial-voice calibration of the ANB-M-C1 in open air by the JRB method (using 115-db signal level) and the MC-253-A was used to obtain the approximate overall characteristic of the microphone-mask combination.

Figure 29 shows a comparison between real-voice and artificial-voice calibration of an A-14 oxygen mask with an MC-253-A microphone. The agreement is fair only at low and medium frequencies. Artificial-voice responses of 12 masks have been compiled in report form.[19c]

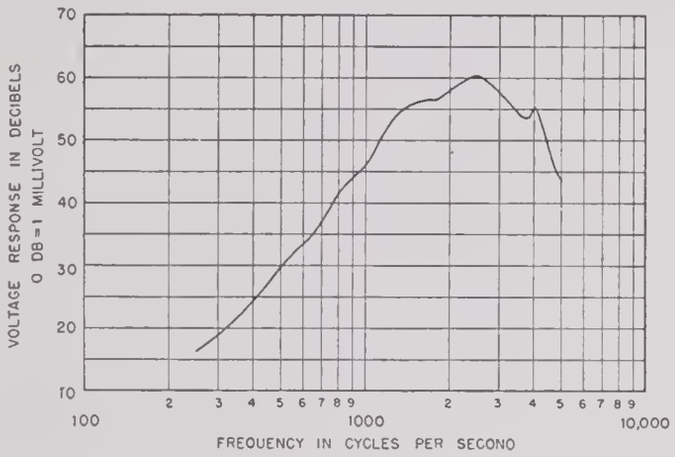


FIGURE 24. Artificial-voice response (JRB) of a type ANB-M-C1 carbon microphone designed for use in oxygen masks. The microphone was measured without the mask.

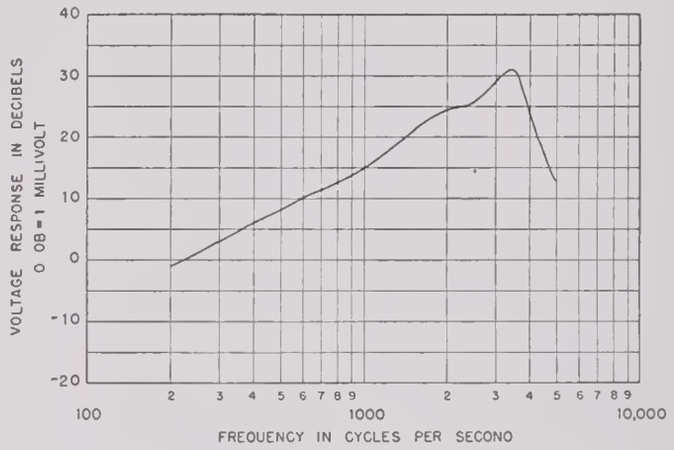


FIGURE 25. Artificial-voice response (JRB) of a type MC-253-A magnetic microphone designed for use in oxygen masks. The microphone was measured without the mask.

10.2.2 Insulation against Ambient Noise

In an effort to provide equipment for rapidly measuring the relative response of microphones to speech and to ambient noise, especially in the case of the noise-canceling (differential) microphone, the Joint Radio Board of the Joint Aircraft Committee established standard procedures in 1944. This equipment consists essen-

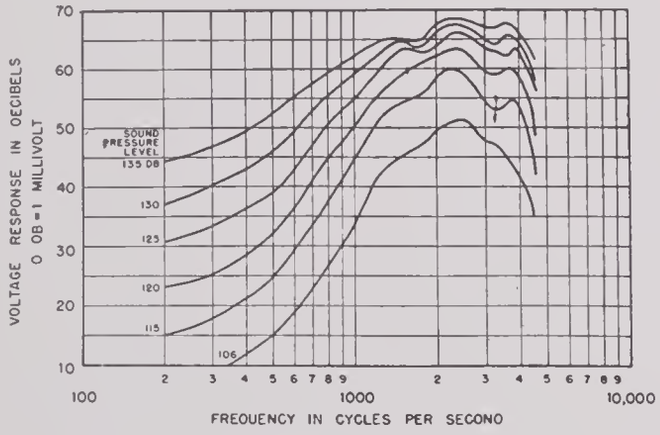


FIGURE 26. Artificial-voice response of a type ANB-M-C1 carbon microphone for various levels of acoustic input.

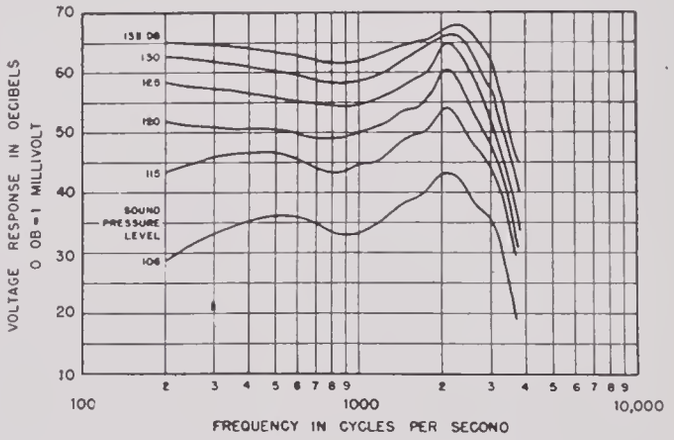


FIGURE 27. Artificial-voice response of a type T-17 carbon microphone for various levels of acoustic input.

tially of an artificial voice of the type described in Section 10.2.1 mounted in a square box lined with absorbing material. "Noise" is introduced from a conventional loudspeaker mounted in one of the walls (see Figure 30). The noise and speech spectra are simulated by line spectra consisting of 10 discrete frequency components suitably adjusted in relative amplitudes. The method and equipment are described in detail in a report published by the Electro-Acoustic Laboratory.[38b]

The method is primarily useful for production testing and to give the design engineer relative estimates of performance. For valid results, speech-to-noise ratio measurements with live talkers are necessary (see Section 10.1.2).

FIGURE 28. Harvard artificial voice with dummy head.

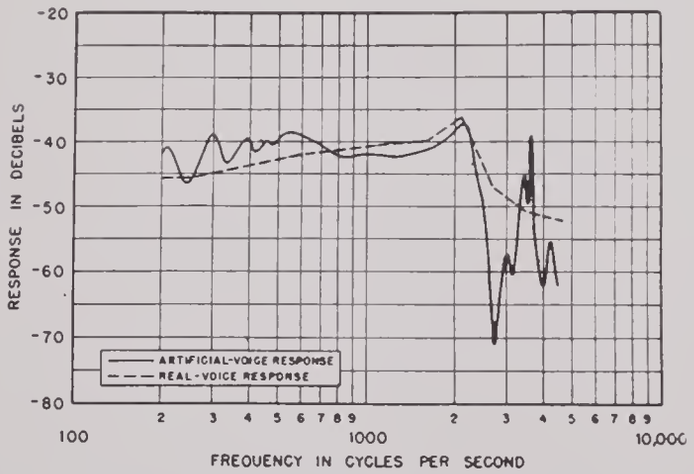


FIGURE 29. Showing a comparison between real-voice and artificial-voice response of an MC-253-A microphone in an A-14 oxygen mask.

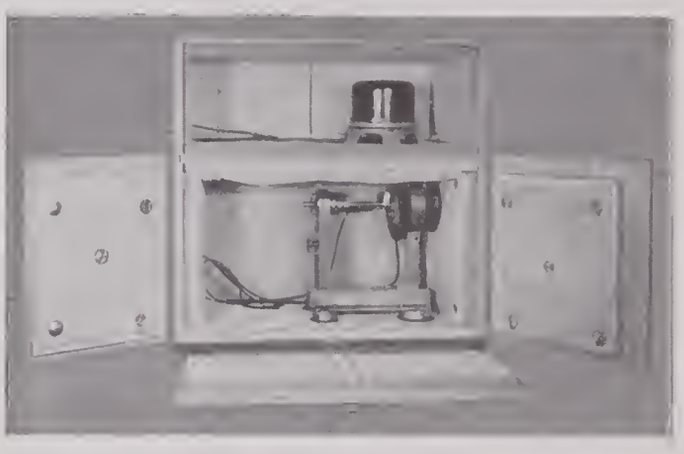


FIGURE 30. Photograph of artificial voice and noise box for testing noise-canceling lip microphones.

10.2.3 Nonlinear Distortion

To measure nonlinear distortion, a source of sound is necessary which generates high (120 to 130 db) sound pressures of sinusoidal wave form at the microphone. An artificial voice of the type already described does not fulfill these requirements. It was necessary, therefore, to design a special sound source.

It can be shown that a combination of two rigid, coaxial tubes of different diameters and lengths will resonate at a succession of frequencies which are not harmonically related. If such a tube system is driven by a sound source (loudspeaker driving unit) at one of its resonant frequencies and the other end is terminated by the microphone under test, a large sinusoidal sound pressure is built up there. Harmonic components present at the driving unit will not be reinforced by the tube. With suitable tube dimensions, a frequency range from 200 to 4,500 c can be covered.

Operating frequencies are restricted to the resonant frequencies of the system. The output voltage of the microphone is analyzed with a suitable wave analyzer, and the harmonics can be expressed in the customary manner relative to the fundamental. Combination tone distortion can also be measured[19d] if two sinusoidal input signals are used.

10.3 METHODS AND EQUIPMENT FOR TESTING EARPHONES ON THE HUMAN EAR

To determine the performance of an earphone on the human ear, some quantitative estimate must be made of how the ear responds to the sound from the earphone. Fundamentally, this requires measurements of the psychoacoustic type.

10.3.1 Frequency Response

A listener wearing an earphone experiences a sensation of loudness when the earphone is energized electrically. If the loudness experienced by a typical listener is the measure of the response of an earphone, the listener's judgment of, for instance, equal loudness is required in order to obtain a quantitative measurement. It will be recalled that a response measurement of this type was said to be the real-ear response (see Section 9.1.3).

If, however, the sound pressure developed at a specified place under the earphone cushion is taken as a measure of the response, the ear is used only as an (passive) acoustic terminating impedance for the earphone. While measurements of both types have been made, those involving loudness judgments are the more laborious and time consuming.

Real-Ear Response (Loudness-Balance Technique)

The real-ear response of an earphone with cushion and headband was determined in the following manner. The subject was placed in the sound field of a loudspeaker mounted in an anechoic chamber (see Section 10.9). The center of his head was located approximately on the axis of the loudspeaker, which he faced at a distance of several feet. The sound field near the subject's head was essentially that of a plane wave.

The sound-pressure level was adjusted to 84 db at the position to be occupied by the subject for all test frequencies. The observer was then placed in the sound field. He wore a wide-range dynamic earphone (transfer standard) with an individually molded earpiece in his right ear. By means of an oscillator, an electronic switch, and two amplifier channels, sound was alternately generated by the loudspeaker and by the transfer standard. The subject was required to adjust the voltage applied to the transfer standard until the two tones were equal in loudness.

Now, the earphone and cushion under test were placed by the observer on his left ear without disturbing the transfer standard in his right ear. The voltages, determined in the previous step, were again applied to the transfer standard and the tone switched alternately to it and to the earphone under test. By means of another judgment of equal loudness, the voltage required to produce a loudness equal to that of a progressive plane wave of 84 db was determined. This is the real-ear response of the earphone and cushion under test.

The procedure just outlined obviates the necessity of disturbing the earphone while making the judgment. Figure 31 shows the average real-ear response of a wide-range dynamic earphone (Permoflux PDR-10) mounted in a sponge rubber cushion (MX-41/AR) of the supra-aural type. Five observers were used to arrive at the average Figure 32 shows the average real-ear response for a headset of the Navy sound-powered type, using a similar cushion (RCA MI-2045E).

Probe-Tube Technique

Measurements of the sound pressure actually developed under the earphone cushion have been carried out for earphones of ten different types with semi-insert tips and earphones using supra-aural cushions.[36] The sound pressure was measured near the entrance of the ear canal.

A thin, calibrated probe tube, suitably shaped and coupled to a Type 640-AA condenser microphone (similar to the instrument shown in Figure 10, Chapter 15) was attached to the earphone and cushion. The end of the probe tube was pushed through the rubber walls of the cushion and was adjusted to lie in the plane through the face of the earphones and near its center. In the case of the semi-insert type earphones, the end of the probe tube coincided with the plane through the end of the tip. By means of the probe tube the earphone response was determined with the earphone in place on the left ears of several subjects.

Figures 33 and 34 show earphone response characteristics for a wide-range dynamic instrument (Permoflux PDR-10), and a headset of the sound-powered type in use by the Navy during World War II. Figures 35 and 36 show the response of the HS-30 and H-16/U headsets used by the Army Ground Forces at the close of World War II. The bars shown in the figures give an indication of the standard deviation of the observations for both frequency and level.

Measurements of this type are useful in determining such important design features as quality of fit and seal and their variations

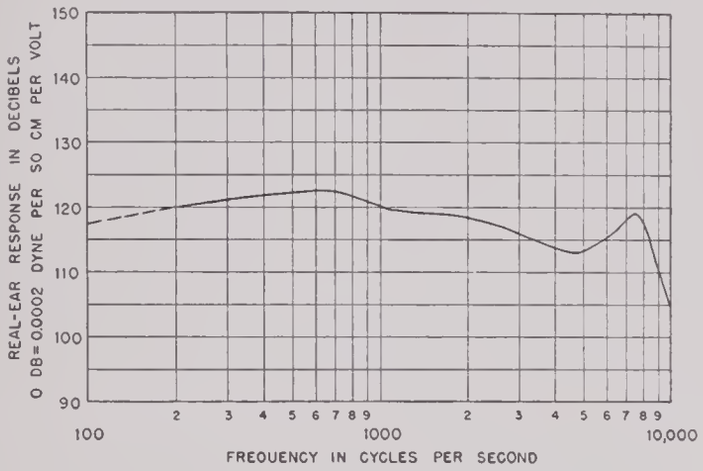


FIGURE 31. Real-ear response of a wide-range dynamic earphone (Permoflux PDR-10) mounted in a sponge-rubber cushion (MX-41/AR). (Constant applied voltage.)

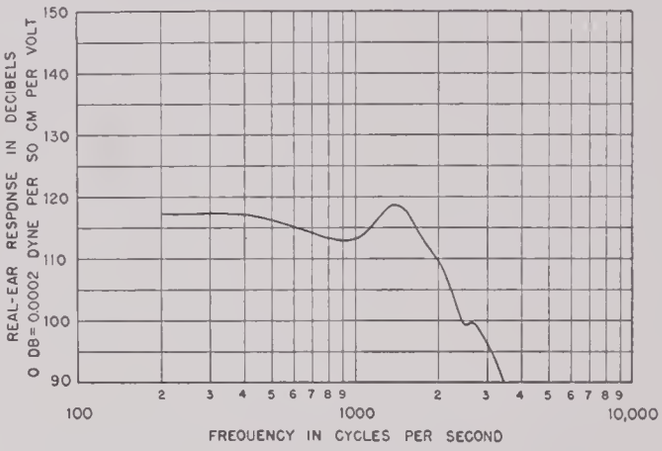


FIGURE 32. Real-ear response of a sound-powered earphone (type M). (Constant applied voltage.)

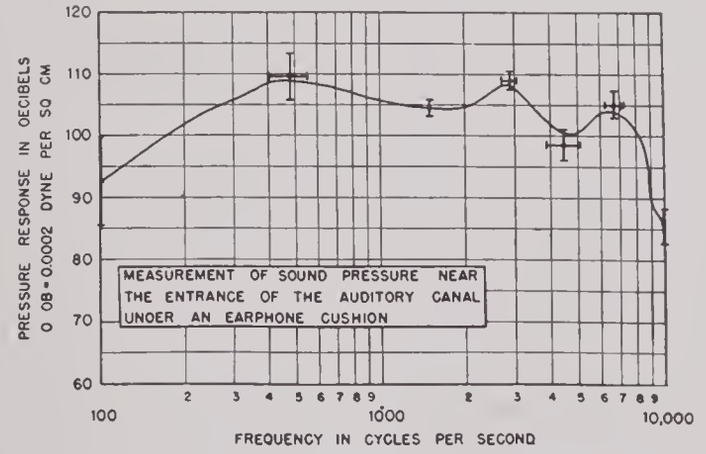


FIGURE 33. Response of a wide-range dynamic earphone (Permoflux PDR-10) mounted in a sponge-rubber cushion (MX-41/AR). (Constant applied voltage.)

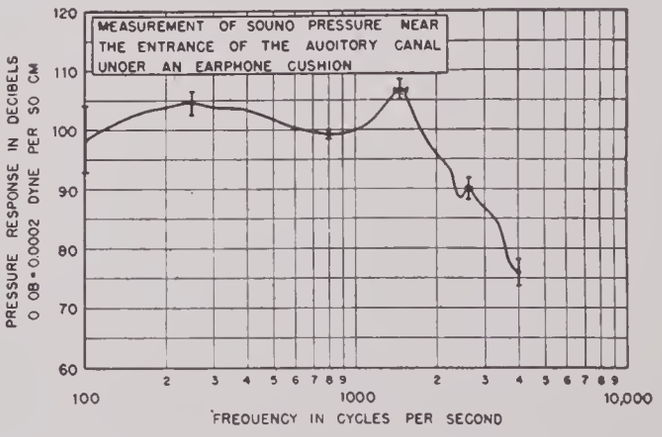


FIGURE 34. Response of a sound-powered earphone (type M). (Constant applied voltage.)

among different subjects. The results also aid in the design of suitable artificial ears (couplers). (See Section 10.4.)

It is interesting to make a comparison of the

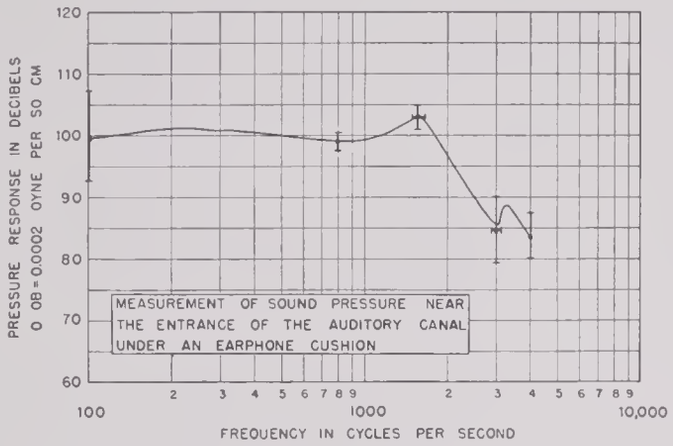


FIGURE 35. Response of a type HS-30 headset. (Constant applied voltage.)

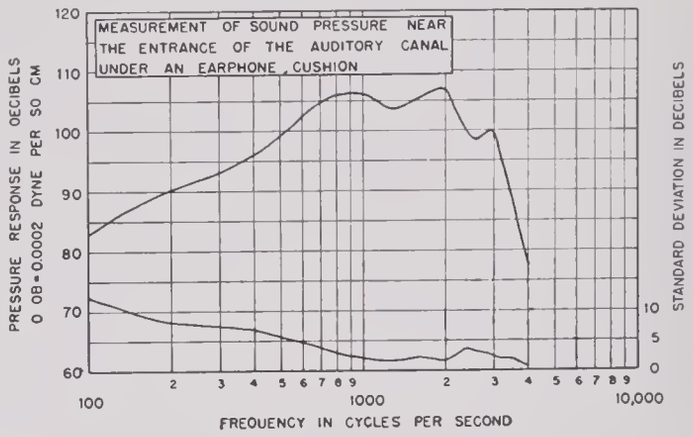


FIGURE 36. Response of a type H-16/U headset. (Constant applied voltage.)

results obtained by the two methods mentioned for a given earphone and cushion. In the response measurement based on a loudness judgment, the ear is used in an active manner as an acoustic indicating instrument. In the response measurement based on the sound pressure under the cushion, the ear is acoustically passive. To compare these two types of measurements, it is necessary to reduce both sets of data to some common reference, such as the sound pressure at the eardrum. To do this, it is necessary to have two further sets of data. First, in order to convert the real-ear response, the sound pressure generated at the eardrum by the sound source in front of the observer must be known for any given free-field pressure with the subject out of the field. To convert the second set of measurements, the ratio of the sound pressure at the eardrum to the sound pressure at the entrance of the ear canal is required. This ratio, combined with data of the type shown in Figures 33 through 36, yields the sound pressure at the eardrum when the earphone is worn.

The first conversion can be accomplished with the information in Figure 3 of Chapter 3. It was observed there that the pressure at the eardrum is greater than the free-field pressure, primarily because of the combined effects of diffraction around the head and pinna and resonance in the ear canal. By combining Figure 3 (see Section 3.1 for azimuth $\varphi = 0$) with Figure 31, Figure 37 is obtained. Figure 37 shows the average sound pressure at the eardrum of a number of observers in a progressive sound field which sounds equal in loudness to the tone generated by the earphone.

Data for the second transformation have been obtained for a dynamic unit (Permoflux

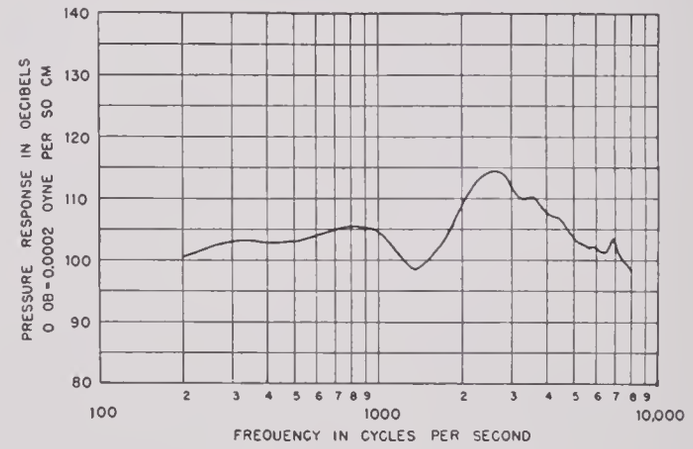


FIGURE 37. Sound pressure at the eardrum generated by a progressive sound field. (Constant applied voltage.)

PDR-10) mounted in an MX-41/AR cushion (see Figure 38) by using the flexible probe microphone described in Section 3.1.[b] By combining Figures 33 and 38, the sound pressure at the eardrum, as measured under the cushion, is obtained (see Figure 39).

[b] The data are only an indication of the effects to be expected and are not generally valid, since only one subject was used.

The sound pressures so obtained (see Figures 37 and 39) at the eardrum might be expected to be equal (within the experimental errors) since the two signals sounded equally loud to

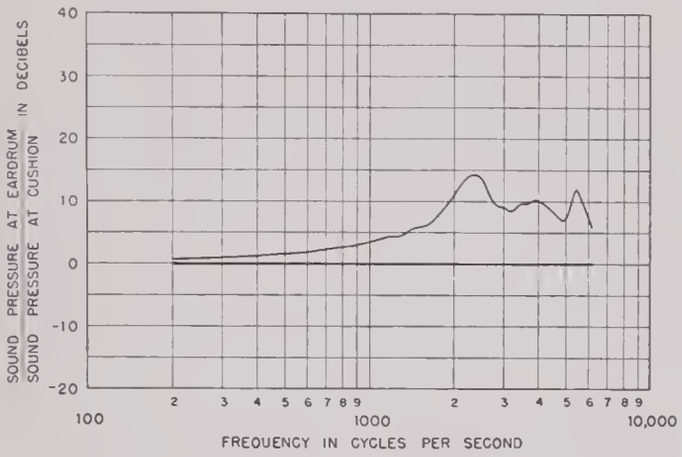


FIGURE 38. Ratio of the sound pressure at the eardrum to that at the cushion for a wide-range dynamic earphone.

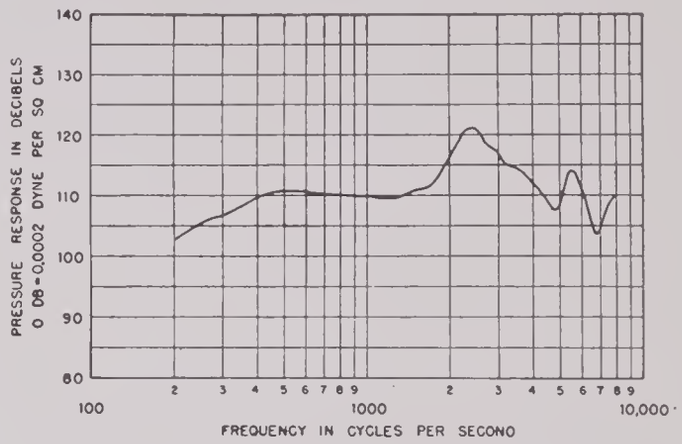


FIGURE 39. Sound pressure at the eardrum for constant applied voltage for a wide-range dynamic earphone (Permoflux PDR-10).

the average observer making the loudness judgment. Actually, the sound pressure generated by the earphone appears to be larger than the one generated by the progressive sound field (see Figure 40). Similar results have been

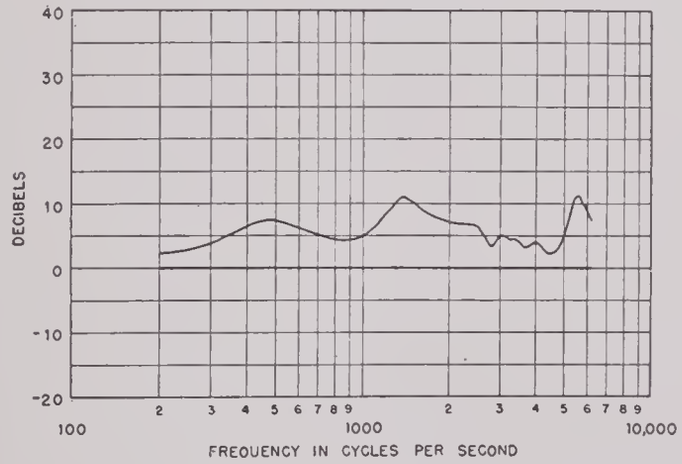


FIGURE 40. Showing the ratio of the sound pressure generated by the earphone to the sound pressure generated there by a progressive sound field sounding equally loud.

obtained for another type of earphone, using a supra-aural cushion, but indications are strong that in this case the results depended to a large degree on the choice of observers.

The disagreement in these results can be interpreted to mean that the ear appraises loudness differently when the signal comes from a remote external sound source than when it comes from an earphone. Similar findings have been reported informally by the Bell Telephone Laboratories. Furthermore, the difference reported[2] at low frequencies between the minimum audible pressure and minimum audible free field (see Section 3.2) appears to be a similar effect and one of the same magnitude.

Measurements on a number of earphones using the insert tips of the type used with hearing aids failed to show the difference discussed above at frequencies below 1,000 c.[29] It must be concluded that the data described here are as yet insufficient to permit general conclusions or explanations, and further research seems indicated.

THRESHOLD TECHNIQUE

There is still a third method of measuring the response of an earphone on the ear by utilizing the auditory threshold. If the threshold for the observer is known in terms of, say, sound-pressure level at the eardrum, a calibration of the earphone can be arrived at. Measurements of this type have been carried out at the Psycho-Acoustic Laboratory[31, 35] for many types of headsets, using a group of young male subjects. The threshold values reported[2] for the minimum audible pressure were assumed to represent the average threshold of the subjects. Figures 41 and 42 show, for two types of headphones using a supra-aural

seal, a comparison between response curves obtained by the threshold method and by the probe-tube method. This comparison suggests the possibility that a uniform shift in the

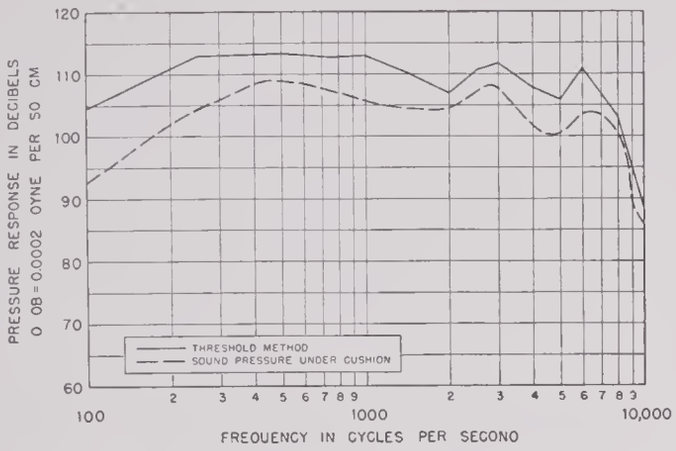


FIGURE 41. Showing a comparison between the response of a wide-range dynamic earphone obtained by the threshold and probe-tube methods.

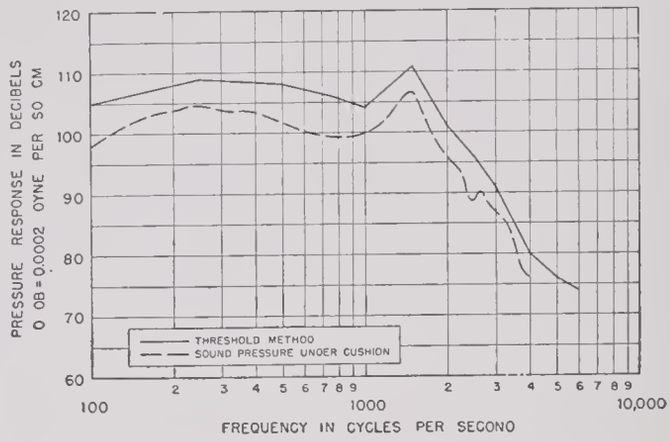


FIGURE 42. Showing a comparison between the response of a sound-powered earphone (type M) obtained by the threshold and probe-tube methods.

assumed threshold would result in better agreement. However, this is not supported by similar comparisons for several other earphones of various types.

10.3.2 Insulation against Ambient Noise

The presence of the earphone and earphone socket shields the ear acoustically from the effects of ambient noise. It is of great importance to measure the insulation (attenuation) provided by various types of earphones and cushions, since the resultant level of noise at the listener's ear is a controlling variable in the performance of an interphone.

Attempts have been made at the Electro-Acoustic Laboratory to measure the insulation by physical means, using a dummy head and replacing the eardrum by a condenser microphone.[38c] Using a slightly different technique, the earphone itself was used as a detector for the sound pressure developed by the noise near the entrance of the listener's ear canal. A condenser microphone connected to a preamplifier by a flexible shielded cable was also used. Measurements were made with single frequencies in an anechoic chamber and with bands of noise in a room with hard walls. The results of these measurements show the necessity, at the present state of the art, of using humans to achieve valid conditions of fit and seal. It is indicated that measurements of the psychophysical type, while more time-consuming and laborious, are to be preferred wherever possible.

Two methods employing human observers have been used at the Psycho-Acoustic Laboratory. One method of measuring the insulation provided by an earphone is based on a direct determination of the masked threshold of the listener wearing the headset under test and immersed in the noise field which corresponds to the conditions of use.[12] Single-frequency tones are used to energize the earphone, and the level is adjusted to be just audible in the presence of noise. From the real-ear calibration of the earphone, the equivalent free-field level of the tones can be found. The equivalent free-field spectrum level of the noise may be assumed to be K db below that value, where K (see Section 3.4.1) is the width of the critical band at that frequency. This spectrum represents the contribution to the total noise at the listener's ear due to earphone leakage.

Simpler, more flexible, and better suited for comparisons between earphones of different types is the method of measuring insulation at threshold in quiet. The threshold method involves the determination of the threshold of hearing at each of 14 frequencies in two experimental situations: first, when the ear exposed to the testing tones is unimpeded, and

second, when it is covered by the headset under test. The difference in decibels between the two thresholds is the amount of protection afforded.

The measurements are made monaurally in an anechoic room, the observer seated with his right ear to a loudspeaker 2 m away. The left

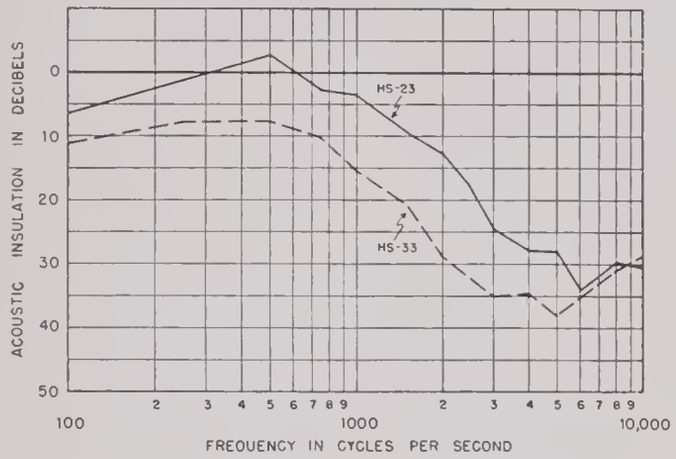


FIGURE 43. Insulation measurements for headsets by the threshold method.

ear is blocked off with an earplug and covered with an earphone socket. The observer sets an oscillator to the test frequency. This tone passes through an electronic switch which interrupts it at irregular intervals and is then amplified and impressed upon the loudspeaker. By the use of a 2-db-per-step attenuator, the observer

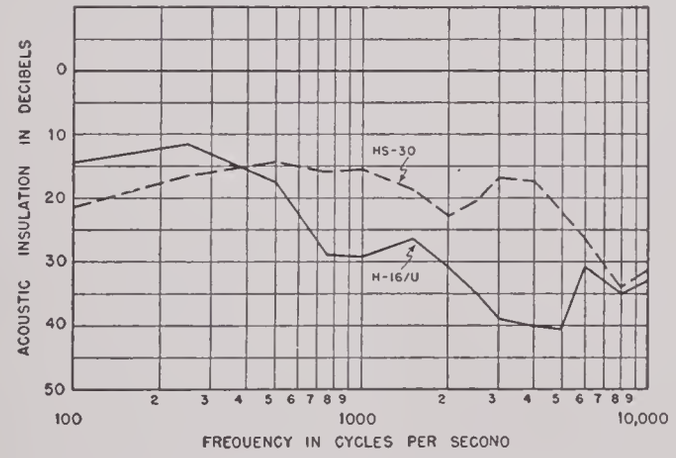


FIGURE 44. Insulation measurements for headsets by the threshold method.

adjusts the intensity of the sound until his threshold is reached.

The observers should be selected for average or better-than-average hearing, and should have considerable experience in determining thresholds with different kinds of headsets before any of their results are used. Experience has indicated that reliable estimates can be obtained by averaging the thresholds for 8 to 10 different observers.

Insulation measurements made by the threshold technique are shown in Figures 43 through 45.[35] Figure 43 shows the insulation

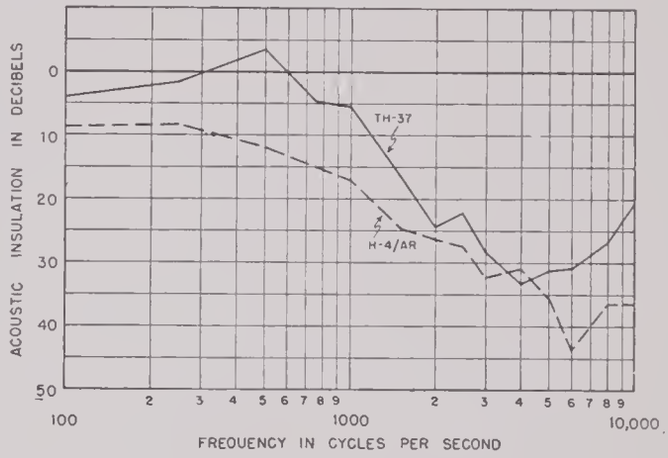


FIGURE 45. Insulation measurements for headsets by the threshold method.

provided by the early AAF headset HS-23 and by the improved AAF headset HS-33. Figure 44 shows the insulation provided by the Ground Forces' headsets. Figure 45 compares the earlier (TH-37) and later (H-4/AR) U. S. Navy headsets. In all cases the adoption of improved cushions increased the amount of insulation provided (see Section 9.4). It will be noted that in all cases the noise exclusion is more effective for frequencies above than below 1,000 c.

10.4 METHODS AND EQUIPMENT FOR TESTING EARPHONES USING AN ARTIFICIAL EAR (COUPLER)

A large amount of work has been done in an effort to replace the human ear in *response* measurements on earphones by a physical model of simple geometrical shape and construction. Tests on such artificial ears (couplers) are far more suitable for the average testing laboratory, especially if production testing is involved. Inherently they are simpler at the expense of validity, as is usual in such situations.

Two couplers have been designed, in coopera-

tion with the Bell Telephone Laboratories, of volumes of 6 cu cm and 2 cu cm, respectively. They are designed to duplicate, as far as is consistent with simplicity of design, the performance of earphones using cushions with a supra-aural seal and semi-insert tips (see Section 11.1). A more complicated design for an artificial ear has been described in reference 1. Both devices are designed for use with the

In 1942, the Joint Radio Board also adopted the design of a coupler of 2 cu cm volume similar to the one shown in Figure 5 in Chapter 15. Subsequent information has shown that this design is suitable only for earphones of the hearing-aid type using *insert* tips.

To investigate the extent to which these two couplers, shown in Figures 46 through 48, duplicate the performance of a given earphone

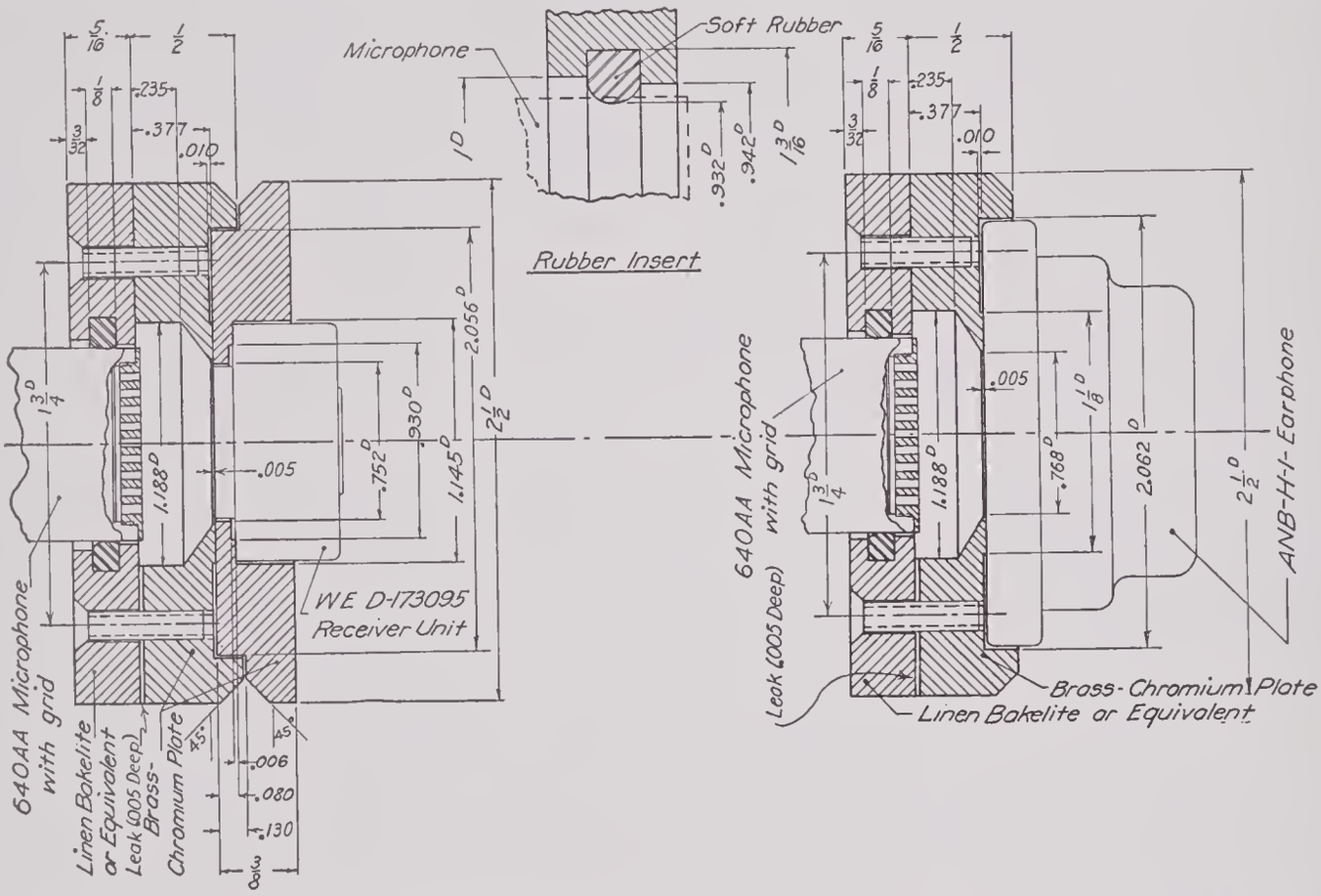


FIGURE 46. Engineering drawings of the standard coupler of 6 cu cm volume for use with the Western Electric type 640-AA microphone with protective grid.

WE 640-AA or 640-A condenser microphone as the pressure-indicating device. The larger coupler was standardized by the Joint Radio Board of the Joint Aircraft Committee in 1942. This coupler consists essentially of a cavity of simple geometrical shape, with means of coupling to it an ANB-H-1 or ANB-H-1A earphone and a condenser microphone. Engineering drawings of the coupler of 6 cu cm volume to be used with Type 640-AA microphones with and without the protective grid are given in Figures 46 and 47. (A drawing of the coupler of 2 cu cm volume for earphones using semi-insert tips is shown in Figure 48.)

on the human ear, measurements of the sound pressure developed under the cushion on the ear (see Section 10.3.1) can be compared with measurements on couplers. Such a study has been carried out for several samples of each of 10 different types of earphones.[36]

Figures 49 through 52 show such comparisons for 4 different types of earphones. From these and other data, the following conclusions can be drawn.[c]

[c] It should be kept in mind that the sound pressure at the eardrum will be significantly higher than the pressure at the entrance of the ear canal at frequencies above 1,000 c.

1. The agreement between the average response in the ear and in the appropriate coupler is fair. If the earphone exhibits a resonant peak in the frequency range between 1,000 and 2,000 c on the coupler, this peak is much reduced in height or not present at all when the earphone is loaded by the human ear. Incorporation of suitable damping elements in the coupler should result in better agreement at tools for many tests on earphones using a supra-aural seal or a semi-insert tip. The performance of cushions of different types, such as the circumaural socket (see Section 11.1) which encloses a comparatively large volume, cannot be duplicated by these designs. Wave motion inside the cavity complicates the design of a coupler having such a large volume. The performance of circumaural cushions is best

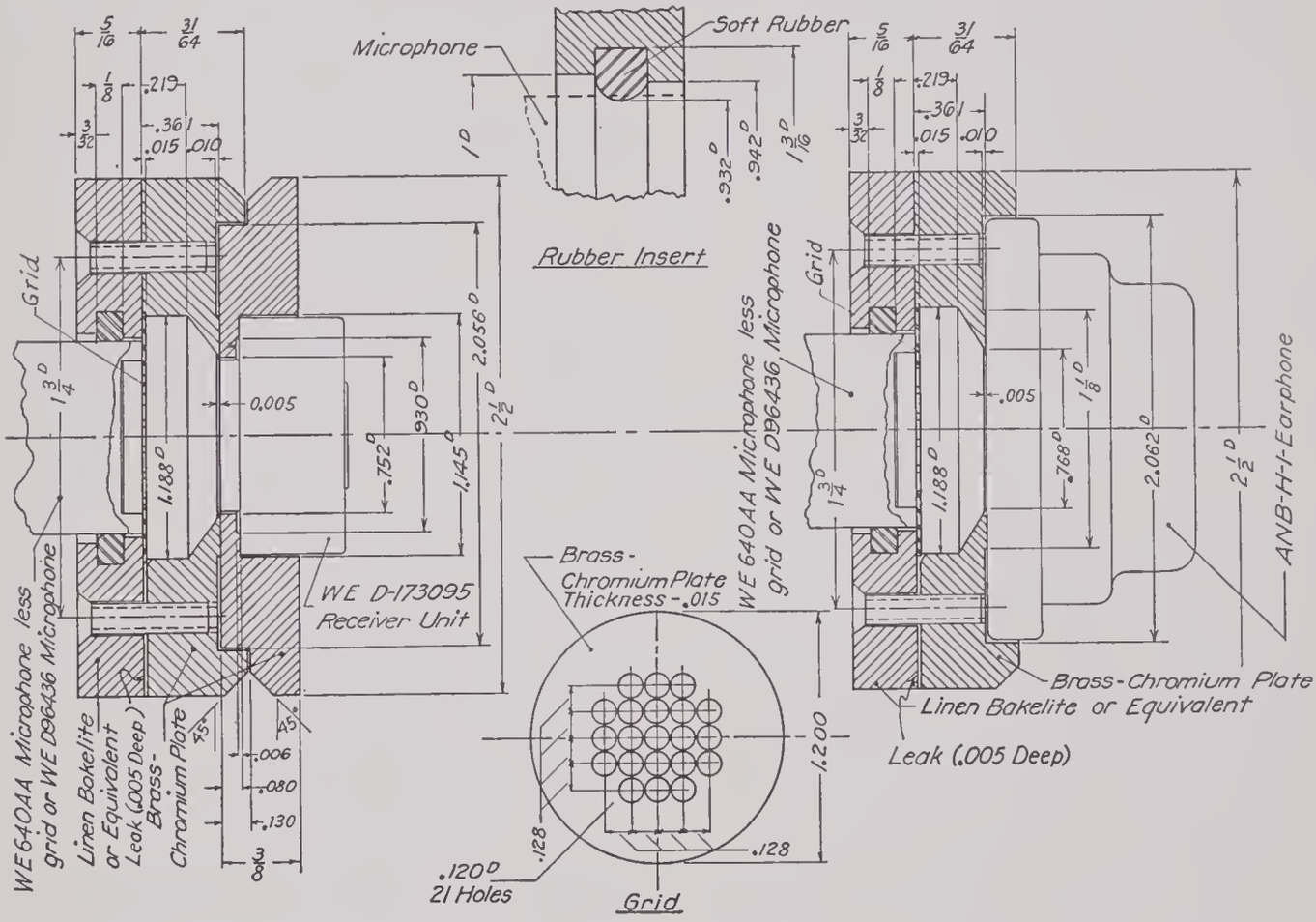


FIGURE 47. Engineering drawings of the standard coupler of 6 cu cm volume for use with the Western Electric type 640-AA microphone without protective grid.

the peak, but this will be achieved at the expense of simplicity and duplicability.

2. The upper cutoff region of the earphone response is reasonably well duplicated by the measurements on the couplers.

3. Leakage effects, resulting in reduced levels at low frequencies and sometimes in elevated levels at medium frequencies when tested on the ear, were intentionally avoided in the coupler design for reasons of simplicity.

It can be concluded that these two designs of artificial ears constitute useful and workable studied by the procedures using human ears, discussed in Section 10.3.1.

It has been assumed, thus far, that the response of an earphone is measured with a constant voltage across its terminals. It is frequently desired to predict the performance of a given earphone energized from an amplifier with a significant output impedance. The impedance of a headset is, in general, complex and a function of frequency. The concept of *constant power available,* developed by the Bell Telephone Laboratories, is useful in predicting the

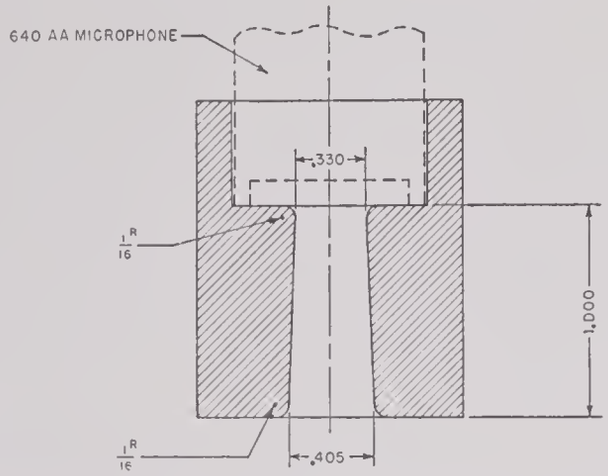


FIGURE 48. Coupler of 2 cu cm volume for use with earphones using semi-insert tips.

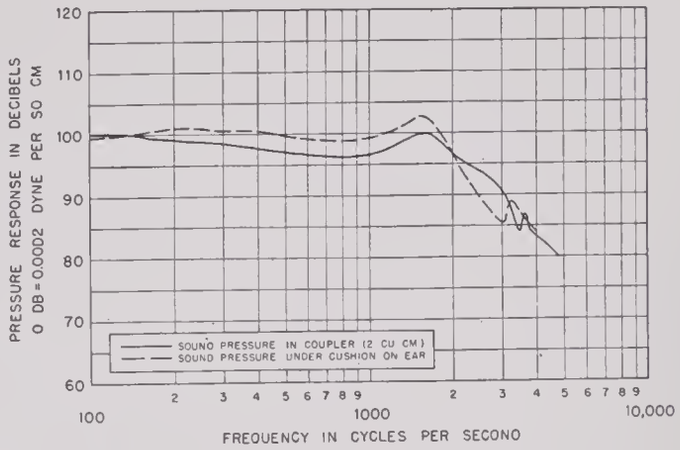


FIGURE 51. Showing a comparison between the response obtained on the human ear by the probe-tube method and the response obtained on a suitable coupler (HS-30 headset).

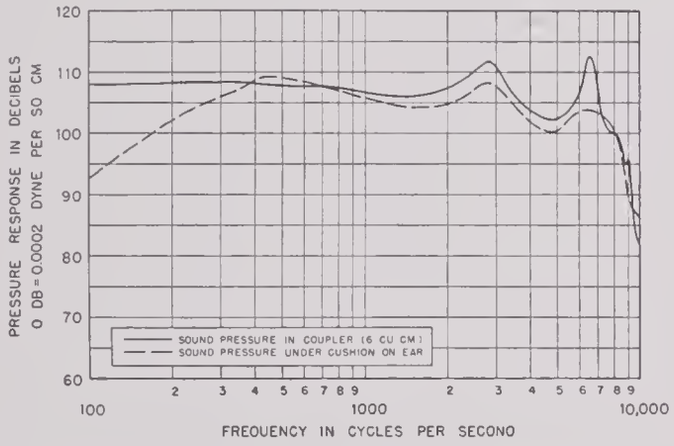


FIGURE 49. Showing a comparison between the response obtained on the human ear by the probe-tube method and the response obtained on a suitable coupler (Permoflux PDR-10).

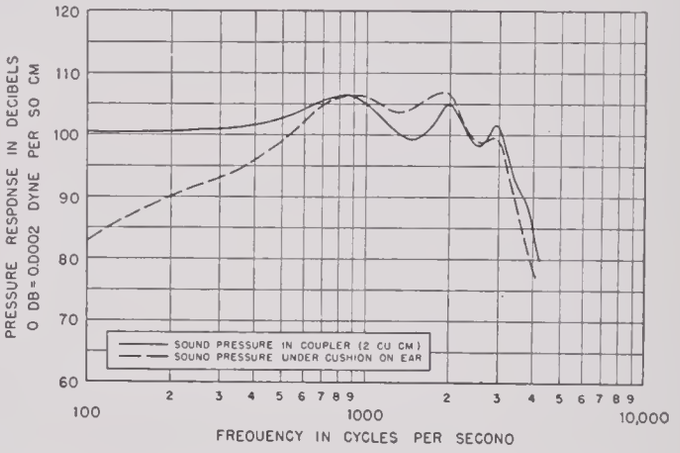


FIGURE 52. Showing a comparison between the response obtained on the human ear by the probe-tube method and the response obtained on a suitable coupler (H-16/U headset).

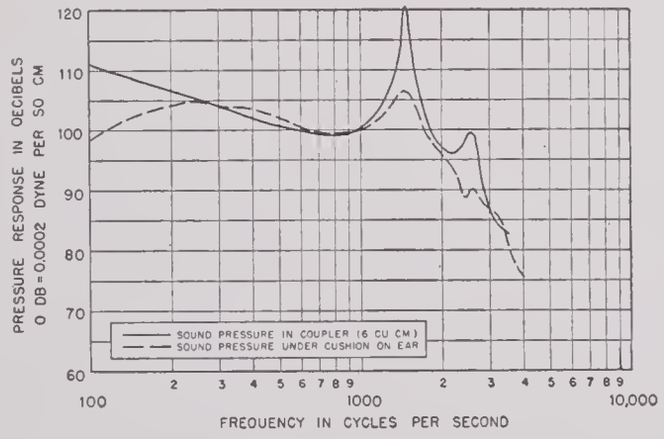


FIGURE 50. Showing a comparison between the response obtained on the human ear by the probe-tube method and the response obtained on a suitable coupler (sound-powered earphone type M).

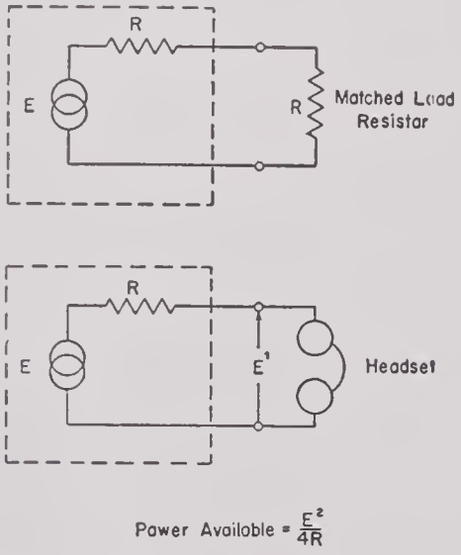


FIGURE 53. Schematic illustrating the concept of available power.

performance and power requirements for headsets when used with sources having a constant output impedance. Consider Figure 53. The amplifier of internal resistance R will deliver maximum power into a resistive load also of magnitude R. In this case the condition of "matched impedance" obtains. The power delivered to the load resistor R is given by $E^2/4R$, where E is the open-circuit voltage of the amplifier output stage. To deliver 1 mw to the load resistor R, a voltage $E = \sqrt{0.004R}$ v is necessary, where E is measured as shown in Figure 54. If a constant open-circuit voltage of

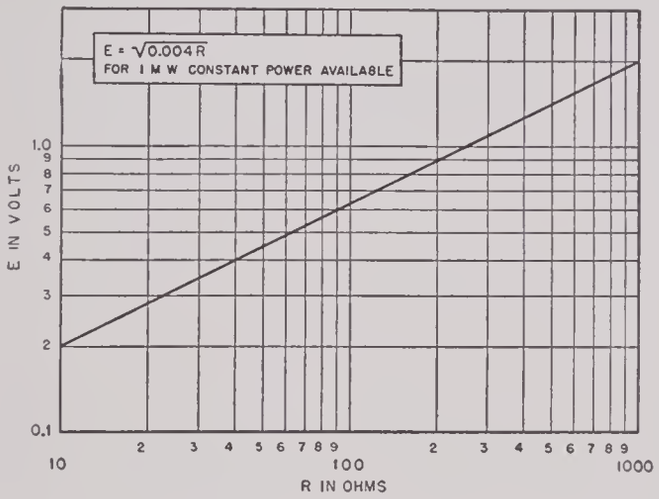


FIGURE 54. Showing the open-circuit voltage E as a function of the resistive source impedance R for 1 mw constant power available.

that value is maintained, independent of frequency, and the headset (earphone) is substituted for the load resistor R, the response measured under such conditions is said to be obtained for 1 mw constant power available. A set of response characteristics using different values of R is valuable in that it permits the selection of that value of R which gives the most desirable characteristic. Conversely, the response of a given headset driven from an amplifier of given internal resistance can be predicted. Measurement of the ratio E'/E (see Figure 53, bottom) permits the calculation of the response characteristic for constant applied voltage. Figures 55 through 58 show the frequency-response characteristics of a number of earphones for 1 mw constant power available, measured on a coupler. Note the significant improvement in the performance of headsets used at the start of World War II (see Figures 55 and 56) and at its close (see Figures 57 and 58).

Couplers can be conveniently used to determine nonlinear distortion of earphones. The sound-pressure meter (see Section 10.6) is suitable for this type of measurement, since it

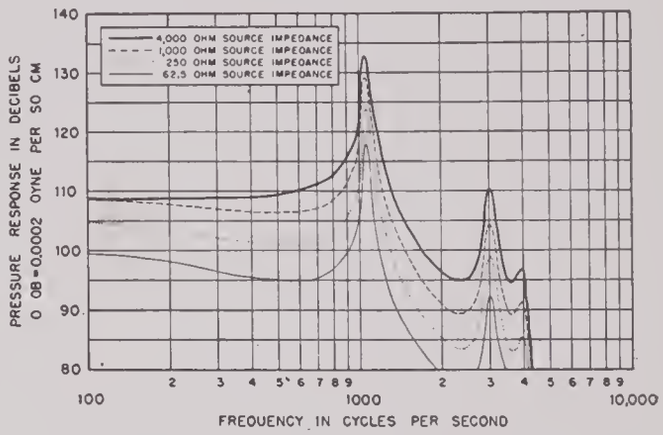


FIGURE 55. Coupler response of an HS-23 headset for 1 mw constant power available.

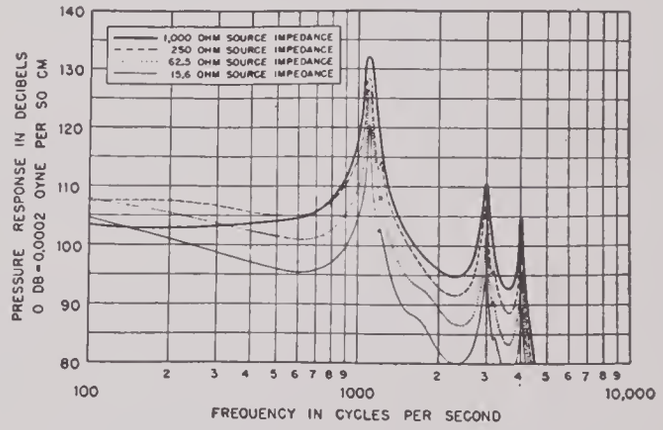


FIGURE 56. Coupler response of a TH-37 headset for 1 mw constant power available.

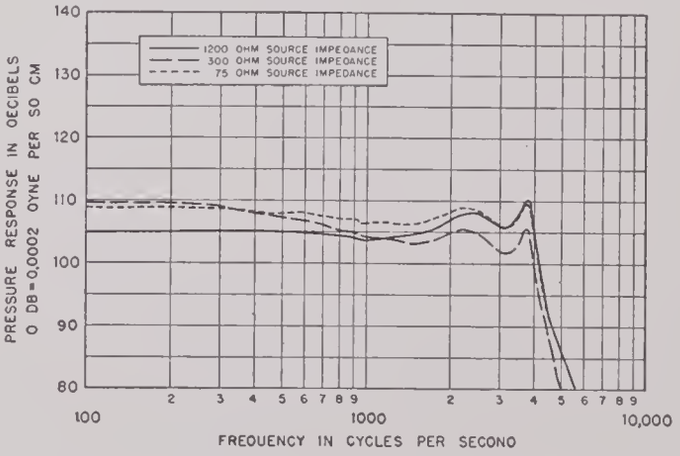


FIGURE 57. Coupler response of an ANB-H-1 earphone for 1 mw constant power available.

is linear up to very high sound-pressure levels.

A large body of physical data on a variety of earphones has been compiled in report form.[19]

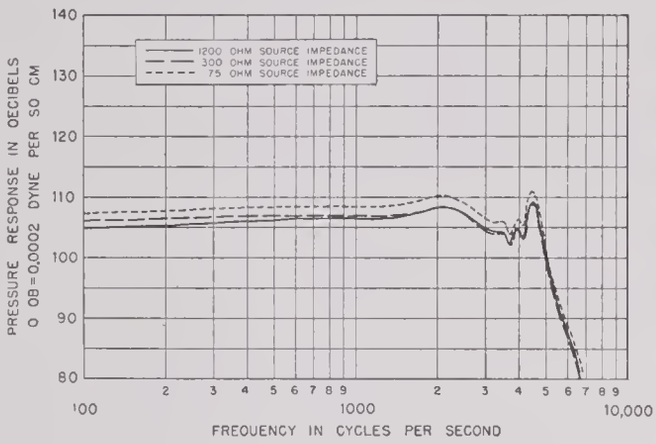


FIGURE 58. Coupler response of an ANB-H-1A earphone for 1 mw constant power available.

10.5 METHODS FOR TESTING INTERPHONE COMPONENTS AT REDUCED AMBIENT PRESSURES AND TEMPERATURES

Tests on microphones in oxygen masks using human talkers at altitude have been reported in Section 9.5. Tests using an artificial voice of the type described in Section 10.2.1 were also carried out, but their validity is extremely doubtful, since the artificial voice fails to react in certain important respects as does the live voice. The altitude decrement of microphones proper can, however, be determined with some confidence by artificial-voice methods.

Direct determinations of the real-ear response at altitude would be difficult. It is customary to express the performance of earphones at altitude by determining the altitude decrement on a suitable artificial ear and assuming that the same decrement would hold on the human ear.

To determine the response of an earphone or microphone at altitude the equipment is placed in a tank in which the desired ambient pressure and temperature are maintained (see Figure 59). For measurements on microphones an artificial voice is used similar to the one shown in Figure 19. To facilitate the test procedure, electric motors are provided to permit rotation of the microphone about its axis and adjustment of its position relative to the direction of gravity from the outside of the tank (see Figure 60).

Most tests have been made at reduced ambient pressure only, since this effect, in general, outweighs the temperature effect. The procedures necessary to secure absolute calibrations of the condenser microphone (which forms an integral part of the test equipment) at reduced ambient pressure and temperature is discussed in Section 10.7.

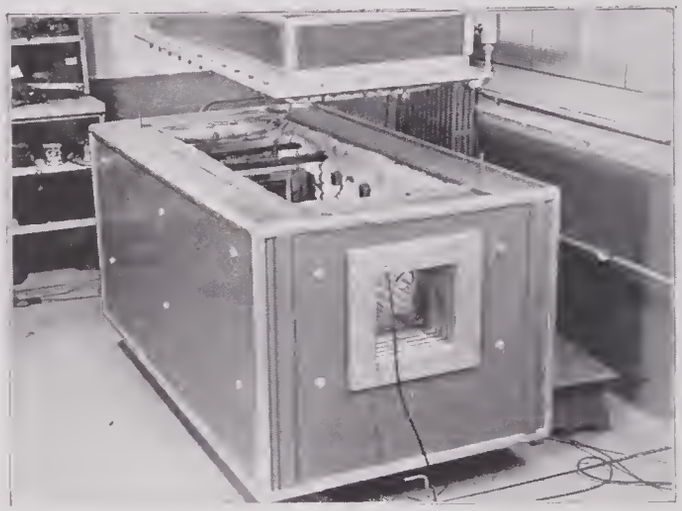


FIGURE 59. Tank for testing interphone equipment at reduced ambient pressures and temperatures.

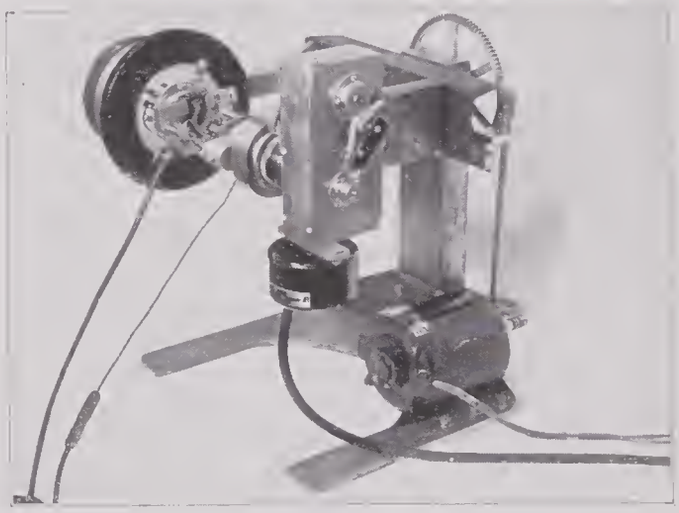


FIGURE 60. Artificial voice with motor controls for use in the tank shown in Figure 59.

Figures 61 and 62 show typical response characteristics for the carbon and magnetic mask microphones ANB-M-C1 and MC-253-A.

10.6 THE SOUND-PRESSURE METER

The sound-pressure meter,[16] incorporating the WE Type 640-AA condenser microphone, was designed at the Electro-Acoustic Laboratory and adopted as the standard instrument for response measurements on microphones and earphones. Numerous specific applications have been mentioned already.

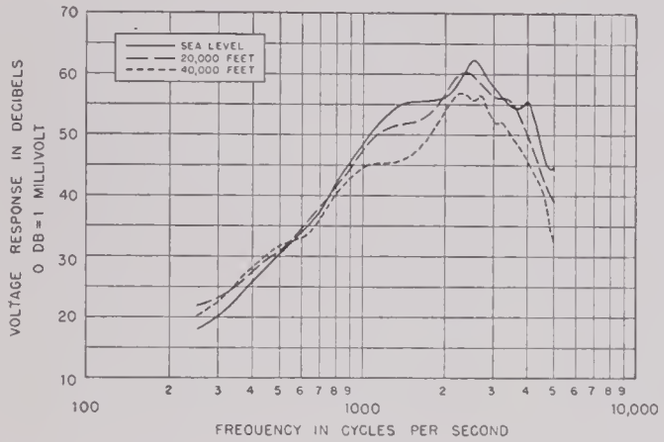


FIGURE 61. Artificial-voice response of a type ANB-M-C1 carbon microphone for various altitudes.

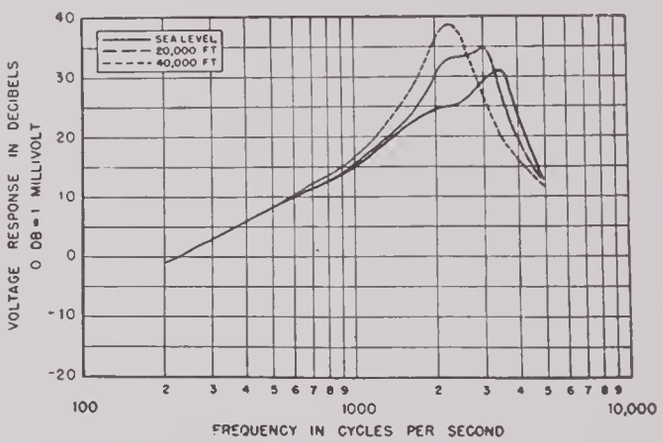


FIGURE 62. Artificial-voice response of a type MC-253-A magnetic microphone for various altitudes.

The instrument consists essentially of a one-stage amplifier whose gain is approximately unity. Its main function is to act as an impedance tranforming device with provision for the necessary bias voltage of 200 v for the microphone. A suitable vacuum-tube voltmeter is connected to the low-impedance output terminals. The input impedance is of the order of 300 megohms. The frequency response is uniform to within ± 0.1 db between 100 and 10,000 c. This is achieved by using a highly degenerative vacuum-tube circuit (cathode follower). It should be noted that the high input impedance is achieved despite the fact that the input resistance for low-frequency fluctuations is only 20 megohms. This substantially decreases hazards due to possible leakage between the terminals of the condenser microphone. The overall noise level is such that measurements of sound-pressure levels above about 80 to 85 db are feasible except in extremely humid weather. The voltage read on the vacuum-tube voltmeter is directly proportional to the open-circuit voltage of the microphone and hence is proportional to the sound pressure.

The amplifier (sometimes referred to as preamplifier) is housed in a cylindrical metal case $1\frac{1}{2}$ in. in diameter and about 12 in. in length. The case is equipped with a thread and contact pin to accommodate the condenser microphone.

Figure 63 shows the preamplifier with vacuum-tube voltmeter. The operating voltages for the instrument are supplied by a 6-v storage battery and a 300-v B battery, or regulated power supply. With proper precautions, the storage battery can be replaced by a filament trans-

FIGURE 63. Sound-pressure meter with vacuum-tube voltmeter.

former. The sound-pressure meter, as commercially manufactured, includes, besides the preamplifier, a shielded battery box and a suitable control box containing the necessary meters. Full details covering operation, calibration, and servicing are available in report form.[16a]

Certain objections have been raised regarding the sound-pressure meter as described above, and a laboratory model of an improved amplifier has been designed.[19e] It combines the best features of the earlier design with decreased noise level, a-c operation on a-c power, reduced size, and potentially higher input impedance. A flexible extension cable of considerable length[19f] connecting microphone and preamplifier can be used. This has advantages in cases where the weight of amplifier and microphone combined is objectionable (see Section 10.3.2). A discussion of the effect of the amplifier input impedance on the noise level of a condenser microphone and ways of further improving the performance are given elsewhere.[19g]

10.7 ABSOLUTE PRESSURE CALIBRATION OF CONDENSER MICROPHONE BY THE RECIPROCITY METHOD

In the early years of World War II, there arose a great need for standardization of test equipment and procedures used by laboratories testing military interphone equipment. Accurate absolute pressure calibrations of the standard microphones were necessary, especially since in many cases the performance of the instruments under test was closely specified.

In most cases, pressure calibrations rather than free-field calibrations were needed. The first step, therefore, was to attempt the standardization of methods for calibrating condenser microphones in absolute terms for use as secondary standards. The condenser microphone most extensively used in this country for acoustic measurements is the Western Electric Type 640-AA microphone. Consequently, all work in absolute calibration of microphones at the Electro-Acoustic Laboratory was concerned with obtaining accurate pressure calibrations of this instrument. The predecessor of the Type 640-AA microphone, the Type 640-A, possesses less favorable stability and temperature characteristics. The diaphragms of these older instruments are made of Dural instead of steel, and are usually coated with blue lacquer for identification. Figure 64 shows a photograph of the 640-AA microphone.

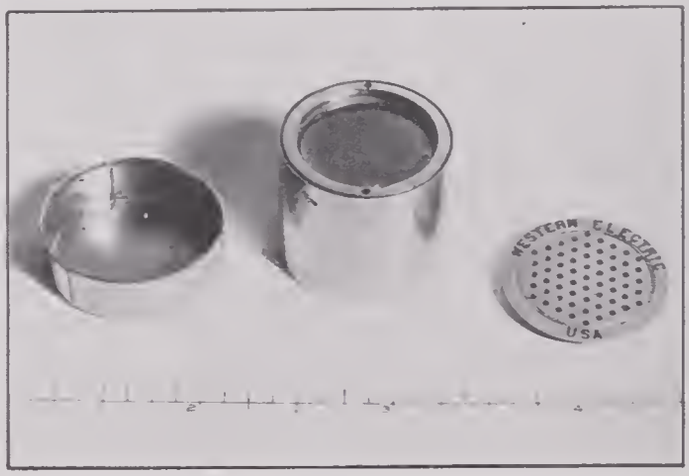


FIGURE 64. Photograph of a Western Electric type 640-AA condenser microphone.

Of the known methods for absolute calibration of microphones in terms of sound pressure uniformly actuating the diaphragm, the reciprocity method has occupied a preferred place in recent years. This method combines accuracy and reliability with simplicity to a degree not equaled by other methods. Microphones with high diaphragm impedance, such as the condenser type, are particularly well suited for calibration by the reciprocity method.

The theory underlying the reciprocity method has been described in the literature.[5, 6] The method rests on the application of the reciprocity principle to a four-terminal network formed by two condenser microphones coupled acoustically by a suitable cavity.

An intensive research program aimed at perfecting and standardizing the experimental procedures was carried on at the Electro-Acoustic Laboratory with the active cooperation of the National Bureau of Standards and the Bell Telephone Laboratories. As a result of these activities, it is now possible to obtain absolute calibrations of WE Type 640-AA microphones accurate to ± 0.1 db in the frequency range of from 50 to 5,000 c and to ± 0.2 db to

12,000 c. These results were confirmed by successive calibrations of a number of instruments at the three laboratories. Figure 65 shows the particular apparatus used at the Electro-Acoustic Laboratory. Figure 66 shows the coupling cavity for the two microphones with preamplifier assembly. (Full details are reported elsewhere.)[37] Figure 67 shows a repre-

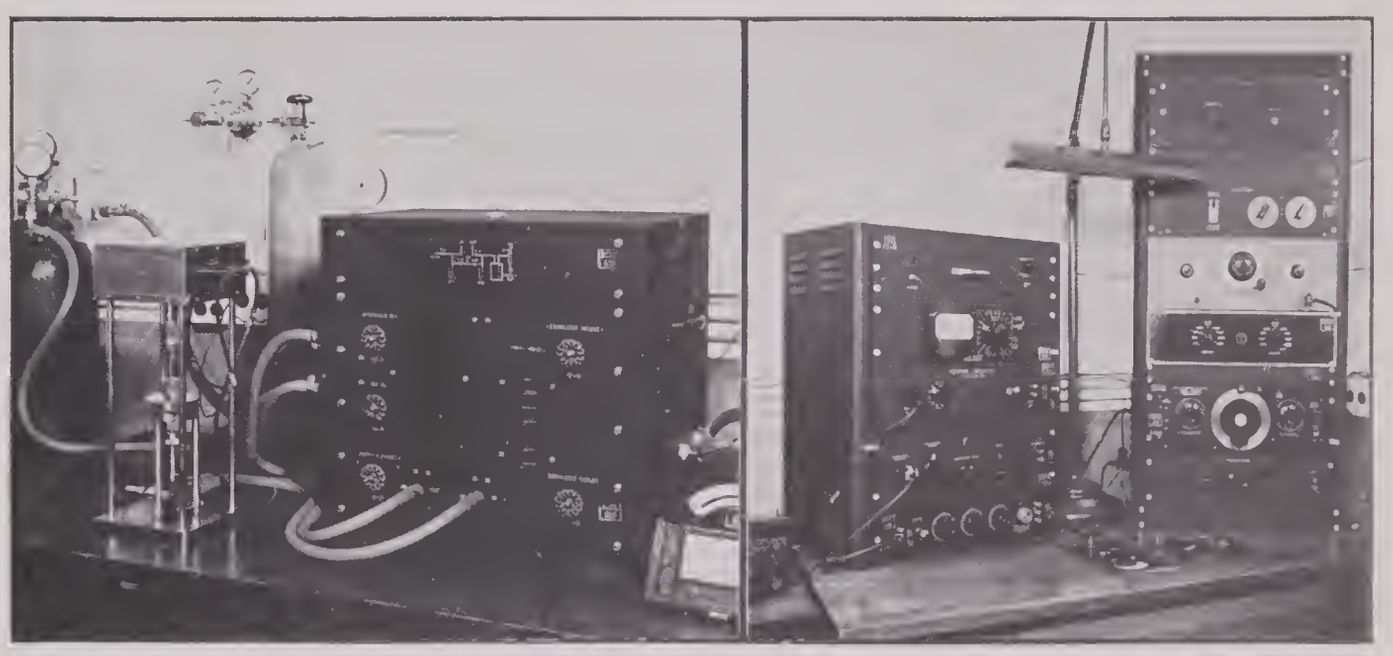


FIGURE 65. Equipment for absolute pressure calibration of condenser microphones by the reciprocity method.

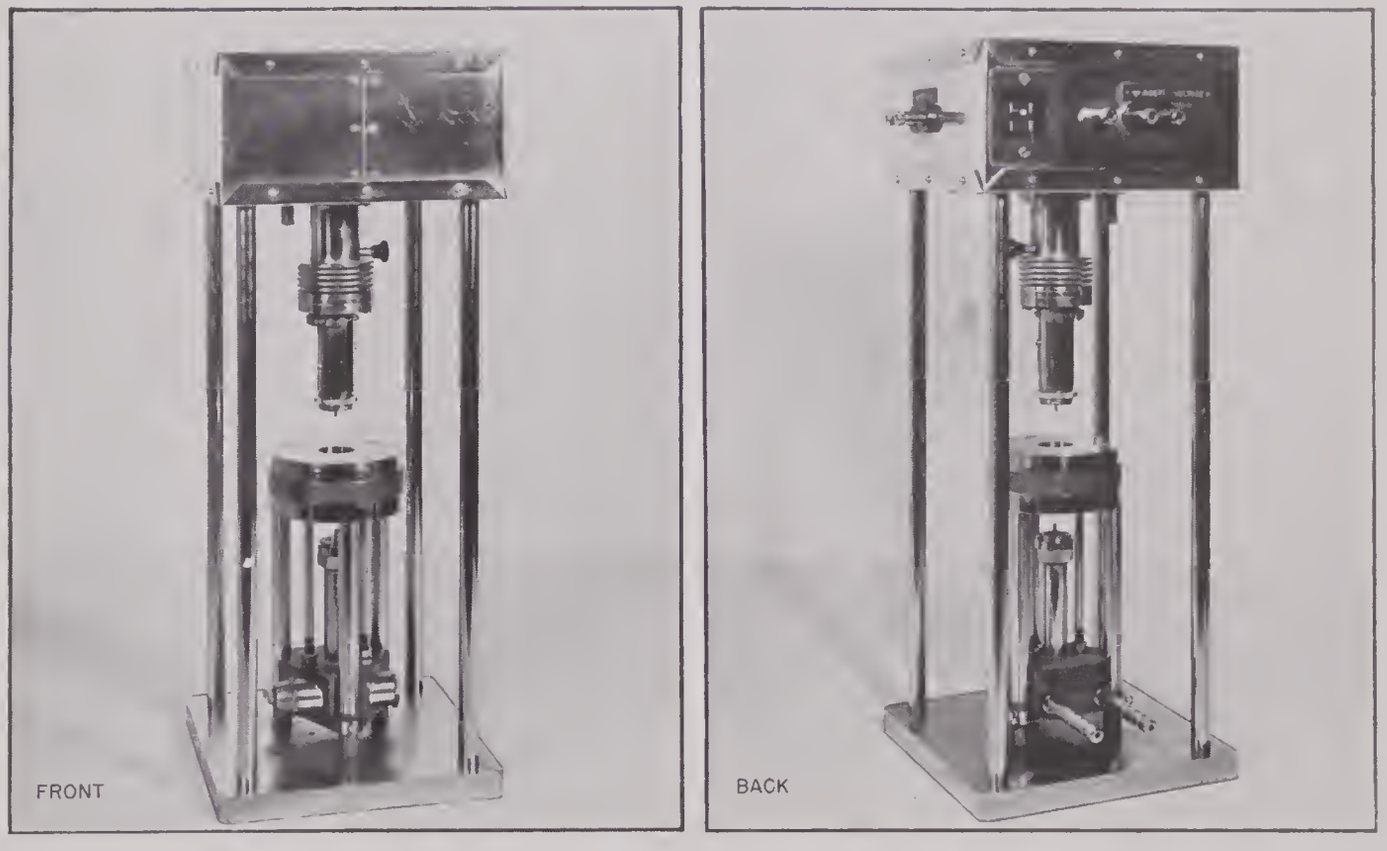


FIGURE 66. Coupling cavity and preamplifier for calibrating condenser microphones by the reciprocity method.

sentative calibration curve for a Type 640-AA microphone.

The frequency range of these measurements has been extended to about 20,000 c. Furthermore, absolute calibrations were carried out at reduced ambient pressures and temperatures.

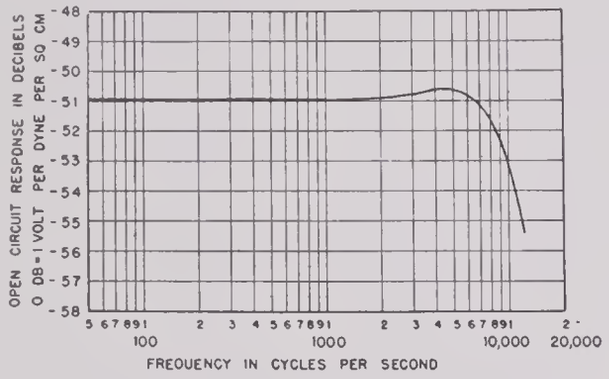


FIGURE 67. Response of a typical Western Electric type 640-AA condenser microphone.

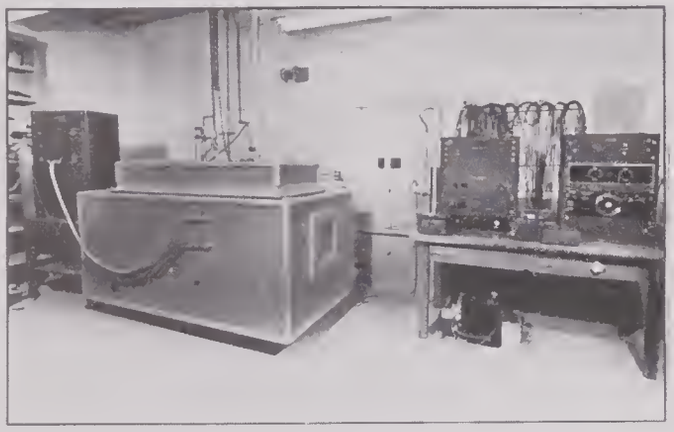


FIGURE 68. Equipment for absolute pressure calibration of condenser microphones by the reciprocity method at reduced ambient pressure and temperature.

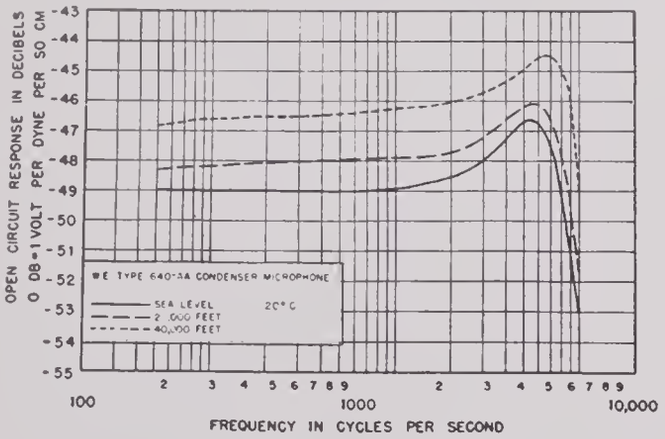


FIGURE 69. Response of a Western Electric type 640-AA condenser microphone at various altitudes.

For this purpose, the coupling cavity with the two microphones is placed in a tank where the required conditions of ambient pressure and temperature are maintained (see Figure 68). Figures 69 and 70 show two typical calibrations of Type 640-AA microphone at altitude and at low temperatures. No measurements were carried out showing the combined effect of low ambient pressure *and* temperature.

10.8 METHODS FOR FREE-FIELD CALIBRATION OF MICROPHONES

As distinguished from the *pressure calibration*, the *free-field calibration* of a microphone is customarily defined as the ratio of its open-circuit voltage to the sound pressure existing at the location of the microphone prior to the insertion of the microphone into the sound field. The sound pressure is produced by a plane progressive sound wave in which the microphone is oriented at some specified angle with respect to the wave front. The ratio of these two calibrations is commonly referred to as the free-field correction.

Free-field calibrations of microphones are required in many practical cases, notably in noise-level measurements (see also Chapter 2).

It is customary, in such cases, to calibrate the microphone in terms of "random" incidence.[4] This calibration is defined in terms of the free-field sound pressure measured by a "point" microphone in a diffuse, or random, sound field. The random incidence calibration can be com-

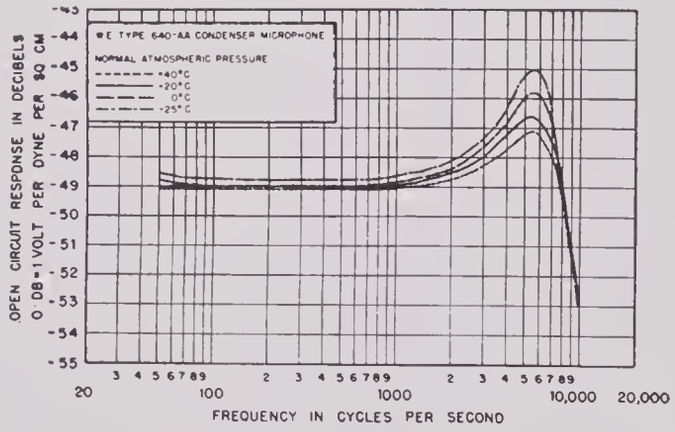


FIGURE 70. Response of a Western Electric type 640-AA condenser microphone at various ambient temperatures.

puted approximately from free-field calibrations obtained in a plane wave for various angles of incidence.[26] It becomes of importance, therefore, to determine the plane-wave free-field calibrations of a number of various microphones for various angles of incidence. These calibrations were obtained by the substitution method, using the Type 640-AA condenser microphone as the standard instrument. The free-field calibration of the standard was obtained by adding the free-field correction to the absolute pressure calibration obtained by the reciprocity method. This correction, shown approximately in Figure 71 for angles of incidence

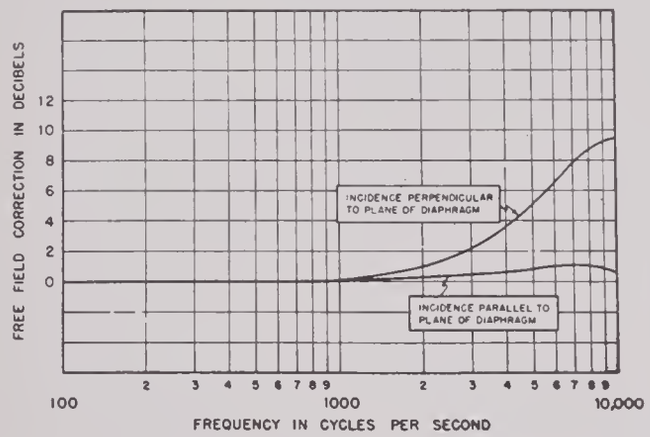


FIGURE 71. Approximate free-field corrections for Western Electric type 640-AA condenser microphones.

normal and parallel to the diaphragm, was obtained from model experiments and from computations of diffraction and cavity-resonance effects. For more accurate work, a direct determination by the reciprocity method would have been preferable.

The free-field correction for random incidence for a 640-AA microphone was also determined and is given elsewhere.[19h] All measurements were carried out in the large anechoic chamber described in Section 10.9.

10.9 THE ANECHOIC CHAMBER

The need for an environment simulating the acoustic properties of unlimited expanse of still air has been apparent in many instances in the course of the preceding discussion. It is hard to underestimate the value of an anechoic chamber, i.e., a room free from reflections, as a research tool in the acoustic laboratory. In order to be effective, the dimensions of the room must be large and the walls must be lined with a sound-absorbing structure which provides maximum and uniform sound absorption over the frequency range in which measurements are to be performed. For accurate measurements, temperature and humidity inside the chamber should be controlled.

A large anechoic chamber was built to meet these needs at the Electro-Acoustic Laboratory in 1943. The design and construction of this chamber have been described in an earlier report.[32]

A photograph of the interior of the chamber is shown in Figure 72. The acoustic treatment

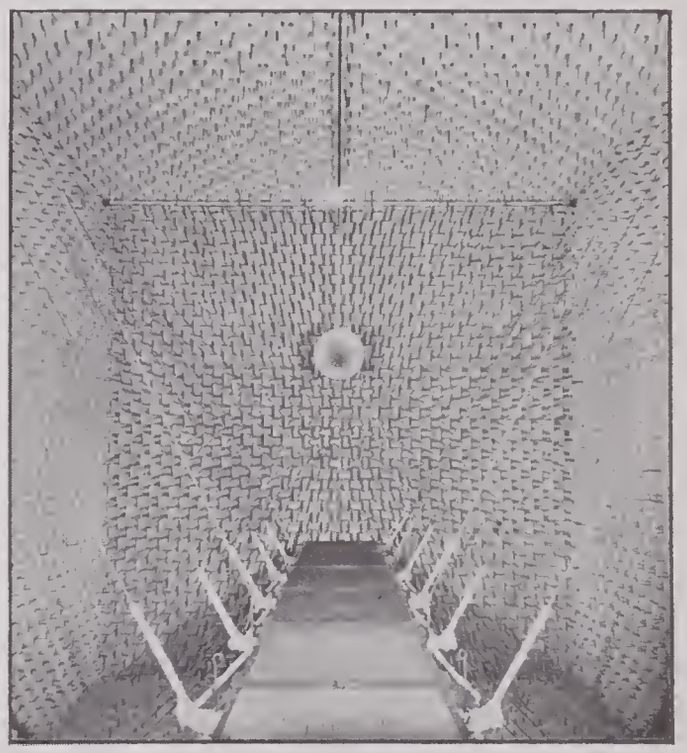


FIGURE 72. Photograph of the interior of the large anechoic chamber at Harvard.

of the walls is of a novel design, consisting of wedges of sound-absorbing material mounted perpendicular to the walls. This treatment has important advantages over treatments of older types[3] because it provides a structure of large depth which absorbs the low-frequency sounds efficiently and at the same time provides for a gradual transition from the acoustic impedance of air to the acoustic impedance of

P.F.-Fiberglas,[d] the material used in the treatment.

The general geometrical shape of the absorbing structure was suggested by a paper[7] describing a room whose treatment consisted of muslin bags of pyramidal shape stuffed with loose rock wool (density about 12 lb per cu ft) and mounted side by side perpendicular to the walls.

This general idea was used as the starting point at Harvard for the design of a better structure. About 500 different designs were investigated experimentally. A structure was finally selected whose main element is a wedge made of P.F.-Fiberglas and contained in a muslin bag. The wedge terminates in a rectangular base section. Butted side by side, this results in the formation of a solid blanket of P.F.-Fiberglas behind the wedges. The bases of the wedges are backed by two layers of Fiberglas blanket, which in turn are separated by an air space from the solid concrete wall forming the chamber. A diagram of the total structure is shown in Figure 73.

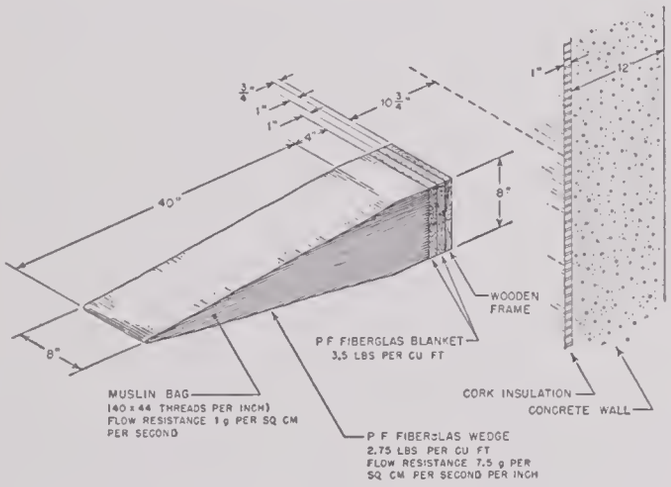


FIGURE 73. Design data and dimensions for the wedge structure used to line the walls of the large chamber.

In order to measure the sound-absorbing characteristics of the structures, the *normal absorption coefficient* α was measured for *one structural unit* by examining the standing wave pattern set up by a sound source in a tube, the termination of which was formed by the element under test (see Section 2.2.3). Instead of using α, the pressure-reflection coefficient can be used to describe the behavior of the structure. It should be kept in mind that measurements of the type discussed above can, at best, yield only a basis for comparison of different structures. The mounting and acoustic conditions in the tube differ so much from those in the chamber that the final performance of the finished structure must also be carefully determined.

Measurements of the pressure-reflection coefficient were made for more than 500 structures of varying design.[e] The designs chosen were limited by the dimensions of the test apparatus at hand[32a] to one single element with a base 8 in. square and to frequencies below 1,500 c. Figure 74 gives the average and limits

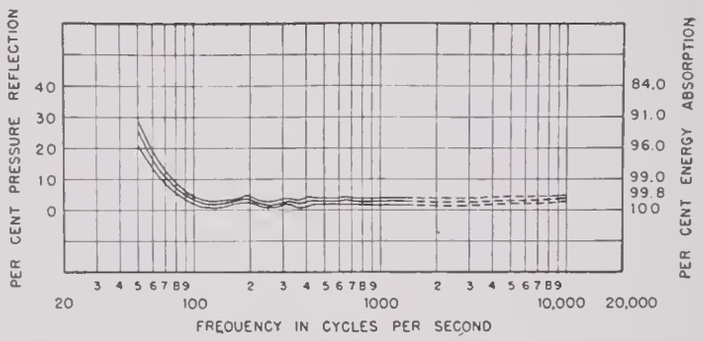


FIGURE 74. Pressure-reflection coefficient for P.F.-Fiberglas as installed in the form of wedges. (Average and limit values.)

of the pressure-reflection coefficient of the structure finally selected and taken from samples as delivered for installation. For this structure, the pressure-reflection coefficient stays below 5 per cent from 100 c up. Measurements in the reverberation chamber show that this design has excellent absorption characteristics also at medium and high frequencies.

From the large body of data on linear wedges of different dimensions[32b] generalized curves were constructed which permit the choice of a good structure if the cutoff frequency is given. This frequency is defined as the frequency at which the pressure-reflection coefficient rises to the value of 10 per cent. Three sets of contours are presented in Figures 75 through 77, relating to the linear dimensions, the flow re-

[d] P.F.-Fiberglas, impregnated with phenol formaldehyde, is a product of the Owens-Corning Fiberglas Company.

[e] Pressure-reflection coefficient, in per cent, equals $100\sqrt{1-\alpha}$ where α is the energy-absorption coefficient.

sistance per unit thickness (see Section 2.2.2), the volume density, and the cutoff frequency. These contours are useful, but it is necessary to use some caution in applying them.

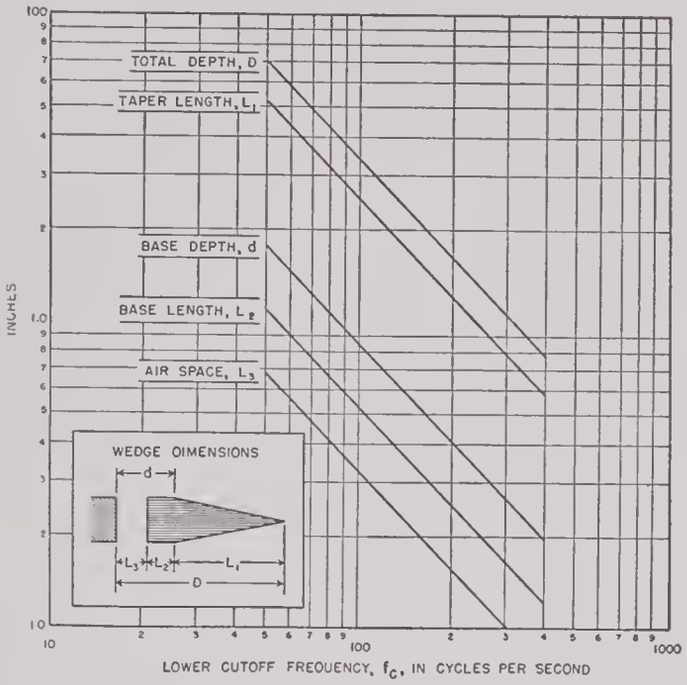


FIGURE 75. Design chart for P.F.-Fiberglas wedges.

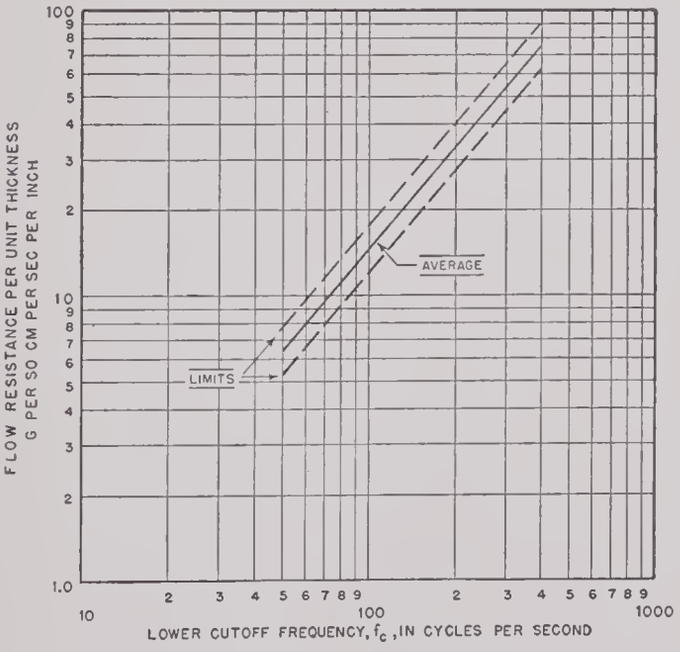


FIGURE 76. Design chart for P.F.-Fiberglas wedges.

Figure 78 shows a sectional view of the large anechoic chamber at Harvard. The inside measurements of the reinforced concrete chamber are length, 50 ft 4 in.; width, 38 ft 4 in.; and height, 38 ft. The floor is 12 in. thick, and the portions of the walls located below ground have a thickness of 2 ft. Above ground, the walls are 12 in. thick.

Spool anchors were provided at intervals of 4 ft to support the framework for the acoustic treatment. Anchors are also provided to hold the cables supporting the two-rail track extending the length of the chamber, approximately 12 ft above the floor (see Figure 72).

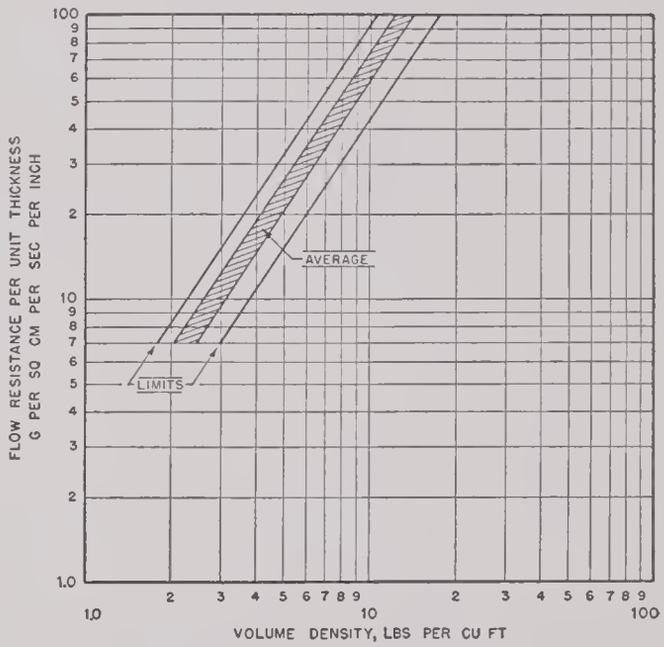


FIGURE 77. Design chart for P.F.-Fiberglas wedges.

A layer of sheet cork was cemented to the walls and floor of the chamber with hot asphalt. This provides for moisture and thermal insulation.

A set of carts can be rolled into the room to provide a floor. For many acoustic measurements the presence of the reflecting surfaces of the carts is undesirable. Rails are suspended from the ceiling so that test equipment can be supported without the presence of the carts.

The treatment is installed by means of a wooden framework anchored to the walls of the chamber (see Figure 79). The air space behind the wedges is broken up by an "egg-crate" structure of Celotex Wall Board. The wooden frames of the wedges are nailed to the furring strips of the supporting framework in an interleaving manner. Supporting chains prevent the wedges from sagging. The edges of

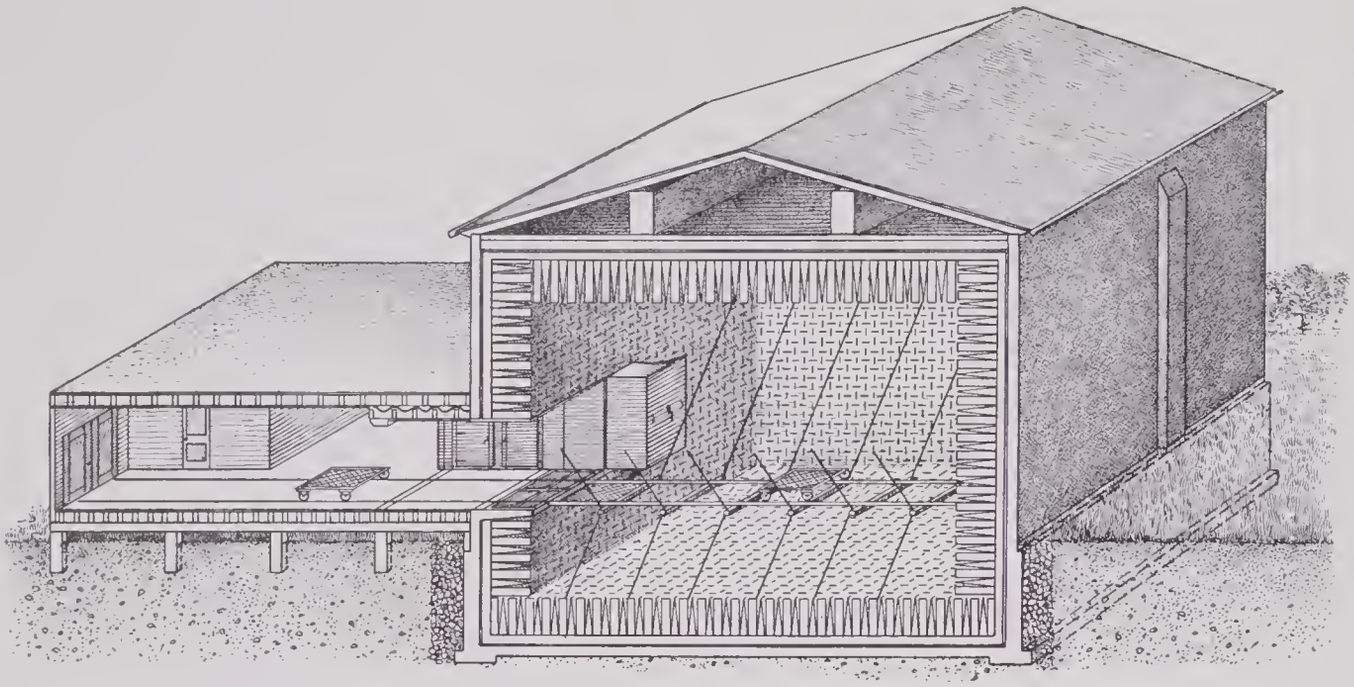


FIGURE 78. Sectional view of the large anechoic chamber at Harvard.

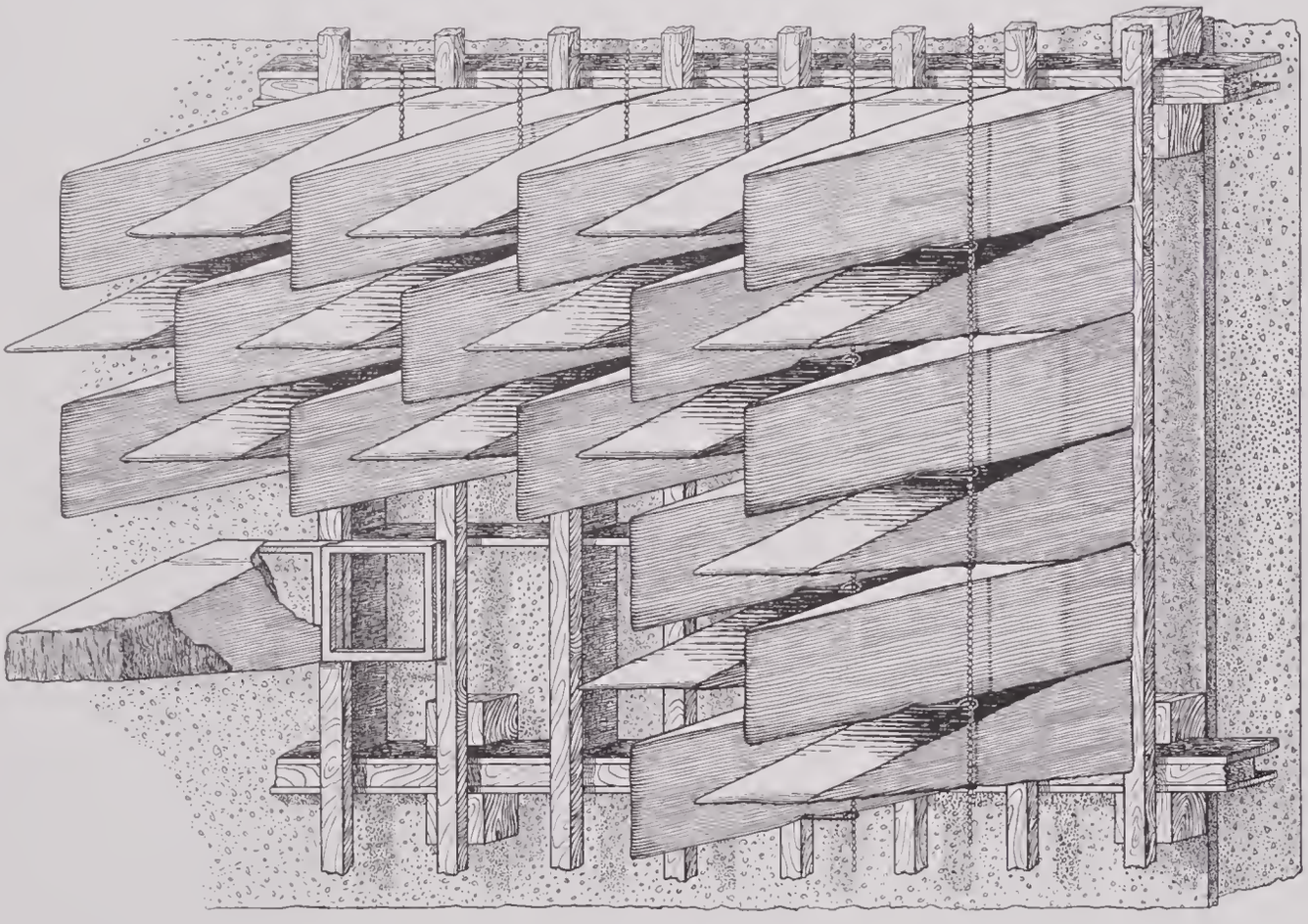


FIGURE 79. Showing details of the mounting of the wedges.

adjacent wedges are at right angles to each other to reduce diffraction effects. Figure 80 shows a portion of the wedges during installation.

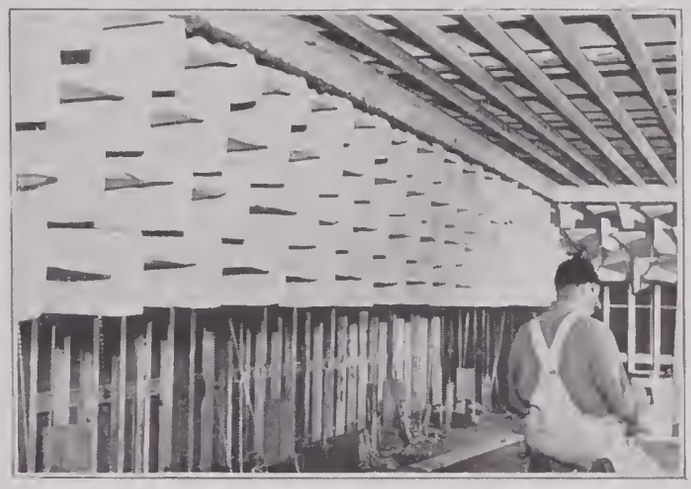


FIGURE 80. Showing a portion of the wedges during installation.

The net working space after installation of treatment is 29x41x29 ft.

As can be seen from Figure 78, only one opening is provided to connect the chamber with the outside. A set of three doors was con-

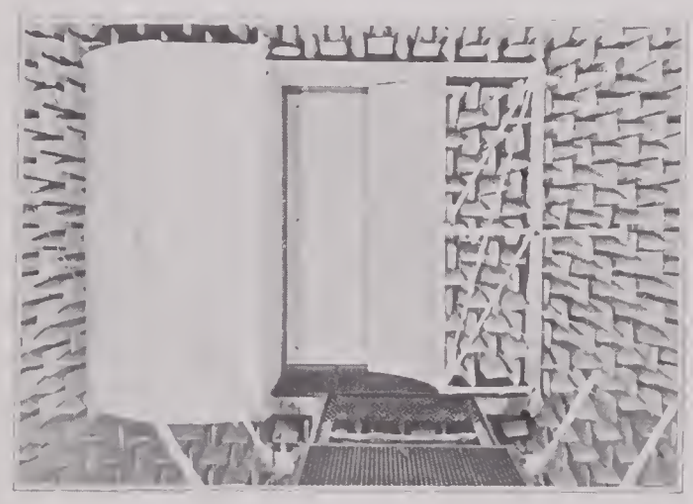


FIGURE 81. Photograph of the inner doors with wedge treatment.

structed to provide an effective acoustic lock. The innermost doors are lined with the same acoustic treatment as the rest of the chamber and are shown in Figure 81. The two outer sets of doors enclose between them a structure separated from the rest of the building by small air gaps. The doors are fitted with heavy hinges, casters, and heavy compression clamps. They consist of sheets of Transite[f] with the interior space packed with soundproofing material. As a result of this careful planning, the overall noise level in the chamber is very low, especially at night when heavy vehicular traffic on a nearby street has ceased.

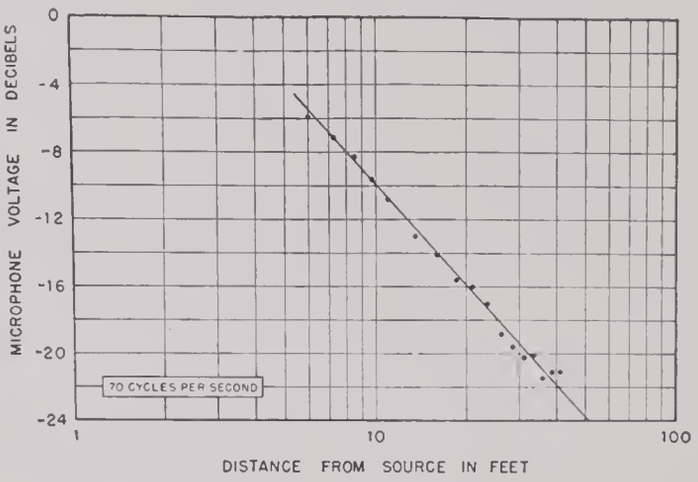


FIGURE 82. Results of measurements of sound pressure vs distance in the large Harvard chamber (70 c).

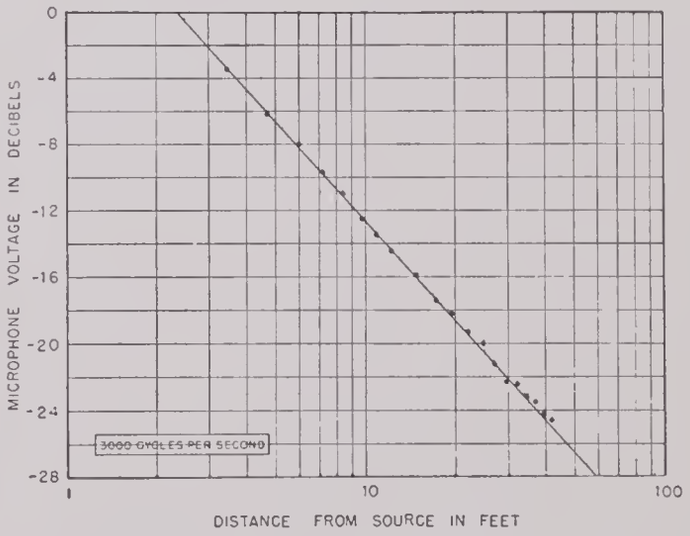


FIGURE 83. Results of measurements of sound pressure vs distance in the large Harvard chamber (3,000 c).

Complete air-conditioning and ventilating equipment is provided. The conditioned air enters the chamber through a removable plug in the acoustic treatment. Air is withdrawn from the chamber through a duct with intakes at the floor level behind the acoustic treatment. Provision is made to remove the intake and exhaust ducts leading from the chamber to the

[f] A product of the Johns-Manville Company.

air-conditioning unit and to seal all openings during measurements.

Although the measurement of the absorption coefficient of the acoustic structure used in the chamber provides one means of assessing the performance of the structure, acoustic measurements in the *finished chamber* are also of

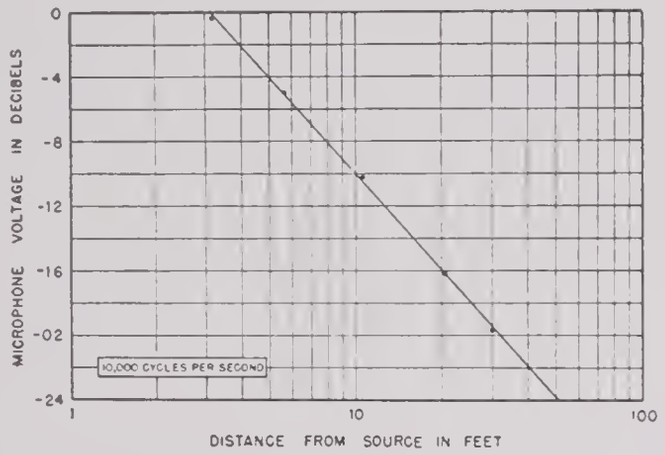


FIGURE 84. Results of measurements of sound pressure vs distance in the large Harvard chamber (10,000 c).

interest. One of the simplest ways of judging how close the chamber approximates free-field conditions is to test the inverse square law. If a sound source is at hand for which the sound pressure varies inversely with distance in free space, measurements of the sound pressure vs

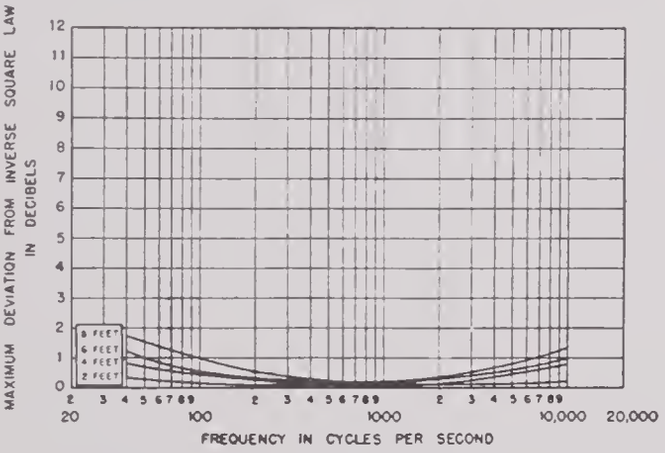


FIGURE 85. Deviations for inverse square law for a commercial chamber.

distance in the chamber under similar conditions will provide a useful index of performance. A large number of measurements at different frequencies for various orientations of source and microphone were taken at the Electro-Acoustic Laboratory. Figures 82 through 84 show a sample of the results of such measurements carried out at 70, 3,000, 10,000 c along one diagonal of the room. It can be seen that even for distances of 10 to 20 ft between

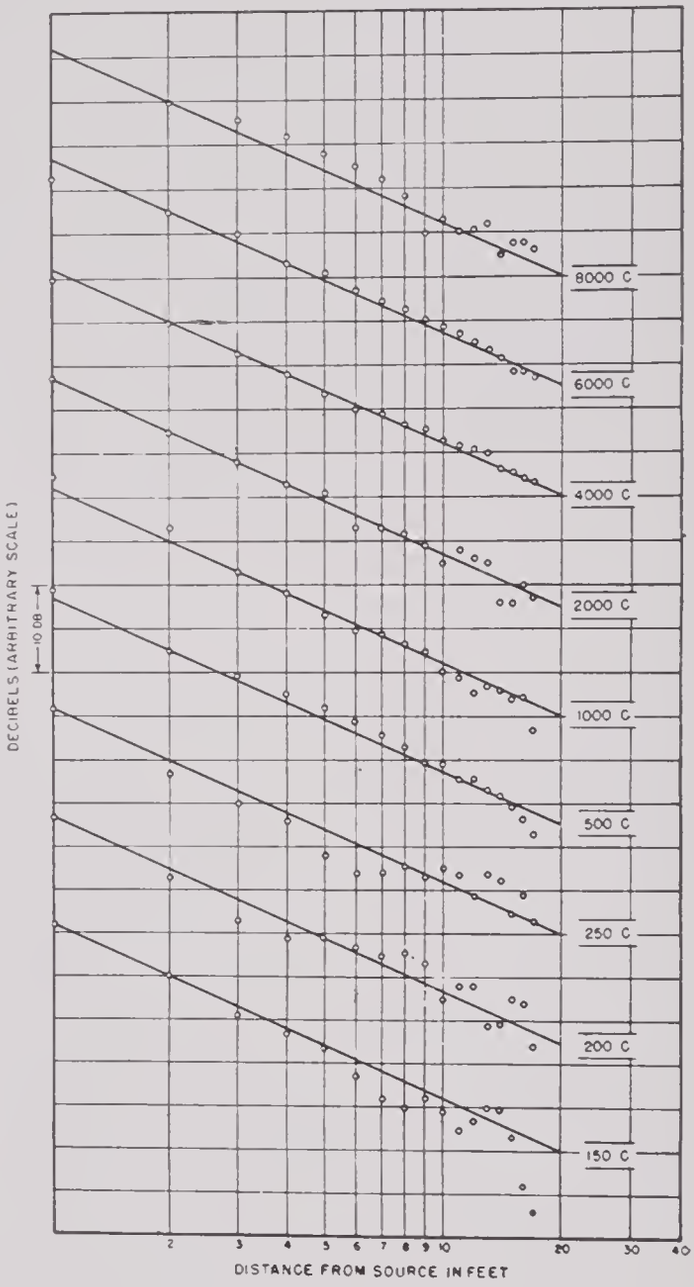


FIGURE 86. Results of measurements of sound pressure vs distance in a chamber lined with wedges at the Psycho-Acoustic Laboratory.

source and microphone, the deviations from inverse square law are small for all important frequencies. These data were taken with all carts removed from the room. For comparison, similar data are given on a somewhat smaller chamber described in reference 8 (see Figure 85).

Measurements of the same type in a much smaller chamber lined with wedges 15 in. long are given in Figure 86. A floor plan and interior view of this chamber, located at the Psycho-Acoustic Laboratory, are shown in Figure 87. The wedges are backed by 2 in. of rock wool, which is, in turn, separated from the wall by an air space of 2 in. Figure 88 shows measurements of the sound pressure vs distance with the rock wool *alone* in place. Comparison with Figure 86 is instructive.

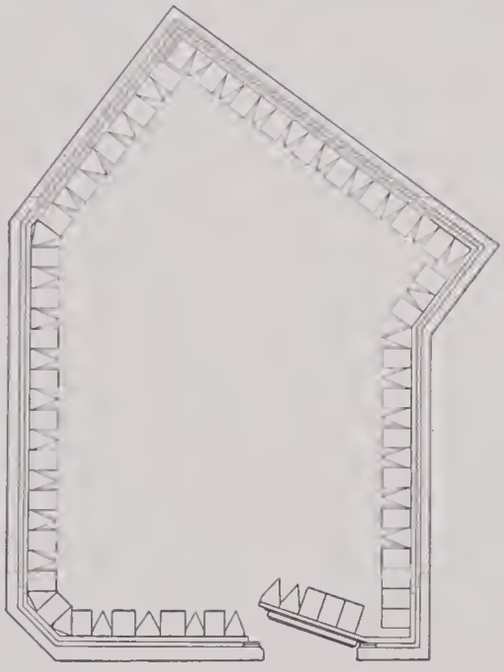


FIGURE 87. Floor plan and interior view of the chamber.

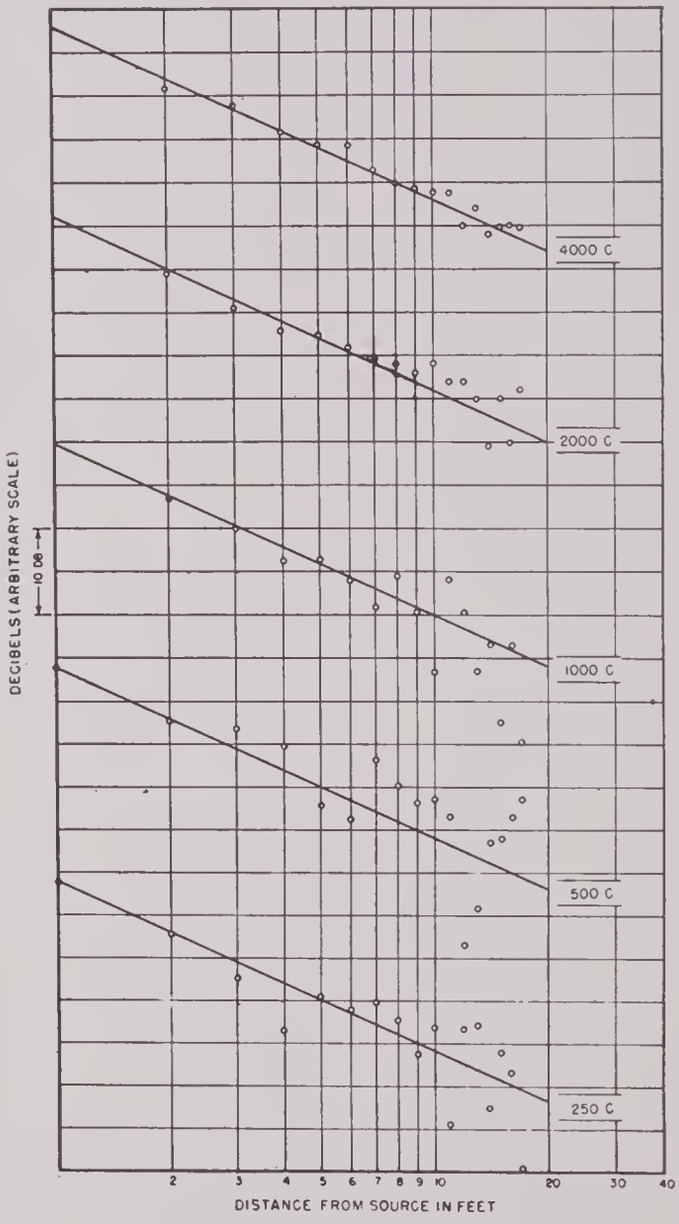


FIGURE 88. Results of measurements of sound pressure vs distance before installation of the wedges. The acoustic treatment consisted of a layer of rock wool.

Chapter 11

THE DESIGN AND DEVELOPMENT OF CERTAIN INTERPHONE COMPONENTS

IT WAS perhaps inevitable that the testing of interphone performance and the standardization of test methods should have led to the development of new components, seeking to improve on the existing designs. At both the Electro-Acoustic and Psycho-Acoustic Laboratories, considerable time has been spent in the modification and improvement of existing communication equipment. But in some cases the entire engineering job was assumed by the Harvard group, an activity which led to applications for more than 65 different patents. The more important contributions made by the laboratories are briefly summarized in this chapter.

11.1 EARPHONE SOCKETS AND INSERT TIPS

All headsets require some kind of a cushion or socket to couple the earphone to the ear. A satisfactory headphone socket must fit closely against the head all around the ear, must be made of sound-attenuating material, must enclose as small a volume as possible, and should be comfortable for the majority of the wearers. Sockets may fit into the ear (insert or semi-insert), over the ear (supra-aural), or around the ear (circumaural), but they must all meet these requirements. Some requirements are easier to satisfy with one type of socket than with another. Comfort is greatest with circumaural sockets, the volume is smallest with insert tips, etc.

11.1.1 Circumaural Earphone Sockets

Earphone cushions developed by the Harvard group were primarily of the circumaural and insert types. One of the earliest projects was concerned with the development of a circumaural socket of the doughnut type (see Figure 31 in Chapter 9) for helmets and headsets.[1] The first design, shown diagrammatically in Figure 1, represents an attempt to use the doughnut construction in a very simple socket. The attachment of the doughnut to the socket allowed the unit to be used in headsets as well as in helmets. Increasing the weight of the rubber backing in order to realize greater protection made the sockets too heavy for comfort. The excessive weight pulled a helmet away from the wearer's head, and larger helmets were required to contain the socket.

Accordingly, Design 1 was altered by reducing its weight and thickness. It was then

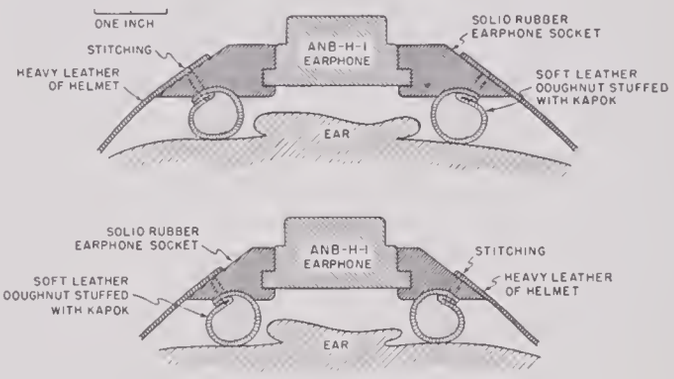


FIGURE 1. Diagram of circumaural socket, Harvard Design 1. Vertical and horizontal sections.

found that this socket, while very comfortable for the wearer, left too large an air volume to be activated by the earphone. Further modifications reduced this air volume, and provided a slight angle to fit the curvature of the head better. This socket, Harvard Design 5, is shown diagrammatically in Figure 2. The slight offset was satisfactory for use in helmets, but it was believed unsuitable for headset use because such a headset required the wearer to distinguish between right and left sockets. Consequently, another socket, Harvard Design 6 (see Figure 3), which would fit on either ear was developed for headset use.

These earphone sockets, when compared with most of the other sockets available at that time, afforded a substantial improvement in insulation without any sacrifice of comfort. The sockets were modified in minor details to suit

specific applications,[9] but the general design was adopted and used in large quantities by both the Army and Navy.

One of the inherent disadvantages of circumaural sockets is the fact that they enclose a large volume of air, a factor counterbalancing their comfort and convenience. Harvard Designs 5 and 6 suffer in this respect, especially as they do not realize the maximum reduction of the air volume possible with the circumaural type, and further developments were directed toward reducing the volume by filling in the space behind the ear and by changing the plane of the face of the earphone to parallel the general forward surface of the pinna. The result of this work was Harvard Design 8-C,[6] shown diagrammatically in Figure 4. A photograph of these sockets in a Canadian headset is shown in Figure 34 of Chapter 9.

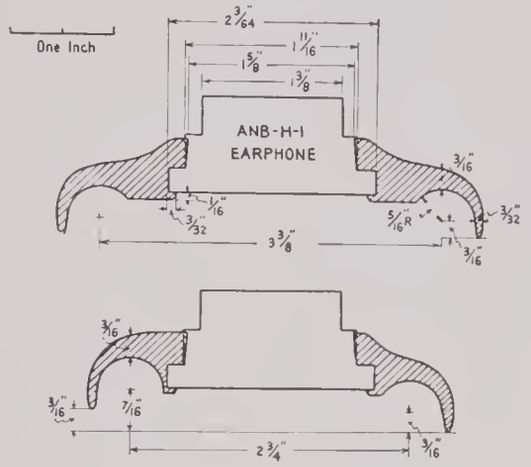


FIGURE 2. Diagram of circumaural socket, Harvard Design 5. Vertical and horizontal sections.

None of these earphone sockets is thin enough to fit under the Type M-1 standard steel helmet, and at the request of the Marine Corps a thin circumaural socket was designed for use with a special thin earphone engineered at the Electro-Acoustic Laboratory (see Section 11.2). The complete headset, adopted for Marine Corps use, is shown in Figure 16.

This socket is interesting in that it is one of the first circumaural sockets used in this country which gives moderately good protection from noise without the use of a doughnut. It is molded in one piece of relatively thin neoprene. The compliance necessary for comfort is obtained by using a relatively soft compound and by making that part of the socket which contacts the head thin and curved. A diagram of this socket, Harvard Design F-3R (Navy designation CW-49506), is shown in Figure 5.

The novel design of the headband also contributed to the thinness of the headset. The usual headband is attached to the back of the earphone and socket, but to save space the headband for the "thin" headset supported the earphone and socket from the side. With this type of socket, the headband pressure necessary for good insulation is less critical than with the preceding types.

11.1.2 Semi-Insert Tips

The development of semi-insert tips presents no fewer problems than the design, construction, and testing of the larger sockets. Neither does the semi-insert involve any principles which differ from those involved in the larger sockets. Instead of being seated around the ear, the semi-insert tips bear upon the surfaces around the orifice of the ear canal. The tip is maintained in position by the headband, as in the case of the larger sockets. The amount of pressure which can be tolerated is, however, considerably less than that used in the larger headsets. The compliance necessary for fitting the semi-insert to the ear is obtained by shaping

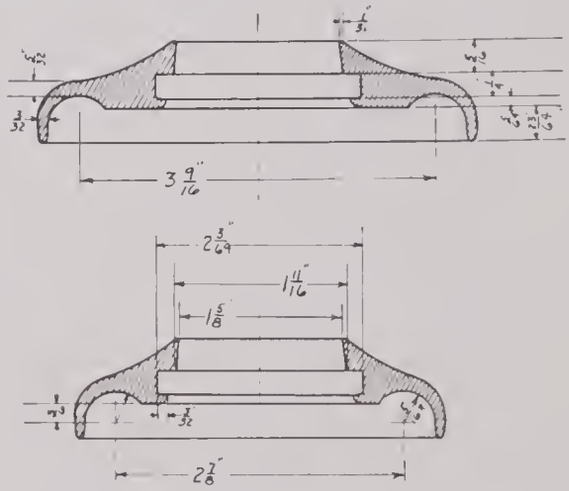


FIGURE 3. Diagram of circumaural socket, Harvard Design 6. Vertical and horizontal sections.

the end of the tip and by adjusting the thickness of the material.

The amount of acoustic insulation afforded by a properly designed and fitted semi-insert tip is far in excess of that which can be expected of any of the sockets previously discussed. The attenuation of ambient sound is more nearly uniform for all frequencies, a condition which is ideally suited for the reception of desired airborne sounds (see Section 2.6) such as the speech of persons near by, while the exclusion of noise is highly favorable to the transduced signal.

The earlier Harvard designs were intended to retain the excellent insulation of the Ground Forces semi-insert socket, M-300 (see Figure 32, Chapter 9), and at the same time to provide much greater comfort for the wearer. One of the successful models, Harvard Design C-6, is shown diagrammatically in Figure 6. The pressure required to produce a seal is very small, and this design is considerably more comfortable than the M-300.

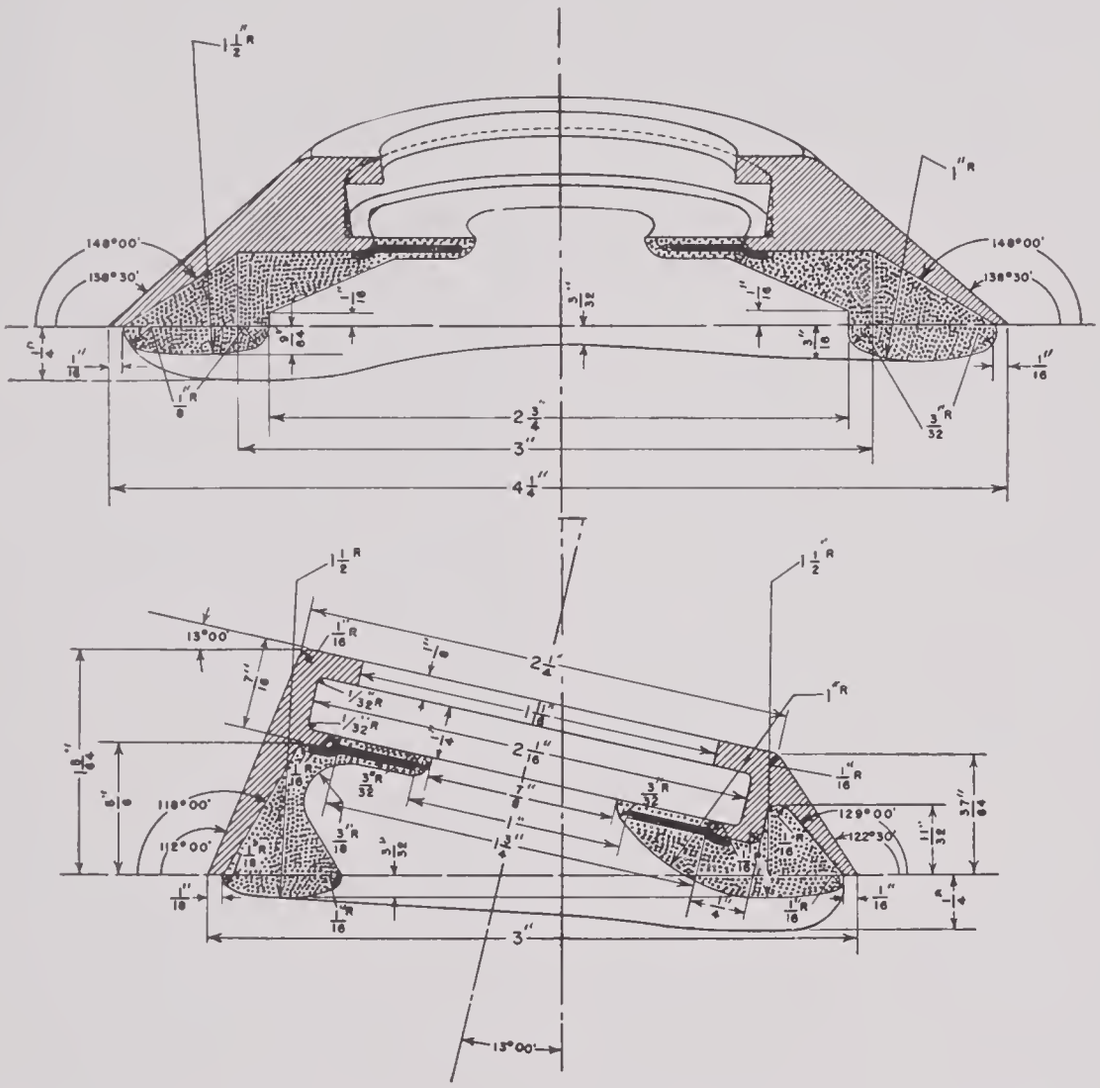


FIGURE 4. Diagram of circumaural socket, Harvard Design 8-C. Vertical and horizontal sections.

11.1.3

Insert Tips

The greatest insulation against ambient noise is provided by the "full-insert" tip. This type of socket is inserted in the ear canal, and supports a miniature earphone without the aid of a headband. It also has the advantage that the

earphone operates into a minimum volume, thereby raising the effective sensitivity of the earphone.

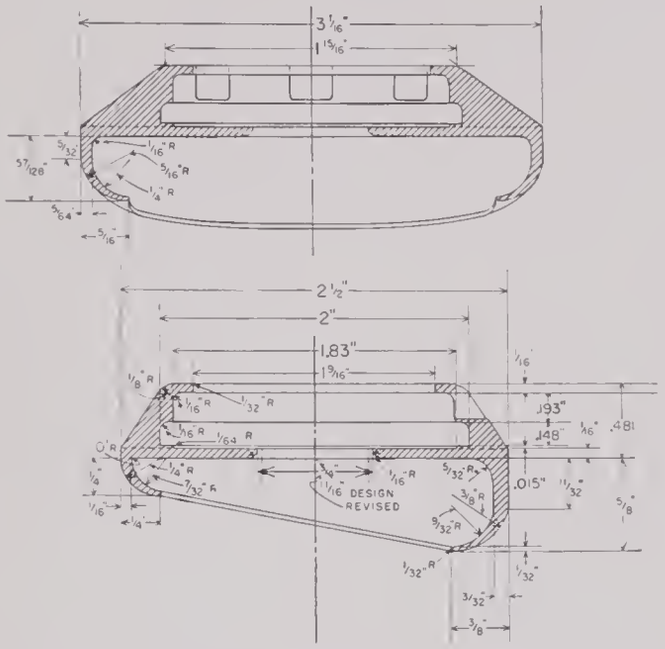


FIGURE 5. Diagram of circumaural socket, Harvard Design F-3R. Vertical and horizontal sections.

A photograph of the Harvard Design M-9 "Harvintip" is shown in Figure 7.

Any device which is inserted into the ear canal creates a special problem in providing a good fit to all ears. Several sizes must be provided, and some care must be exercised in pushing the tip into the ear. Most users can be well fitted, but some observers find the Harvintip uncomfortable. Softer materials are a possible solution to this problem. Sanitary problems are similar to those encountered with earplugs (see Section 2.6).

11.1.1 Dual-Seal Socket

Various attempts have been made to combine two of the generic socket types into a single unit. The Ground Forces headset H-16 U (see Figure 33, Chapter 9) illustrates the combination of the circumaural and semi-insert designs into one socket. At the Electro-Acoustic Laboratory a cushion was developed which combined the circumaural and the supra-aural designs. A cross section of this dual-seal cushion is shown in Figure 8, and a photograph is given in Figure 9. The socket consists of two annular-shaped cushions. The outer cushion is designed to seal against the head around the ear, and the inner cushion approximates a seal against the pinna. By virtue of their thinness and flexibility the cushions may readily assume the shape of the head or ear. Tests indicate that the insulation of this socket is equivalent to that of the doughnut types, and the smaller volume of air which the headphone must activate should increase the sensitivity of the headset. It should be noted, however, that a highly refined design is required in order that both seals be effective on any large proportion of heads. Both of the above designs must be regarded as somewhat tentative in their present form.

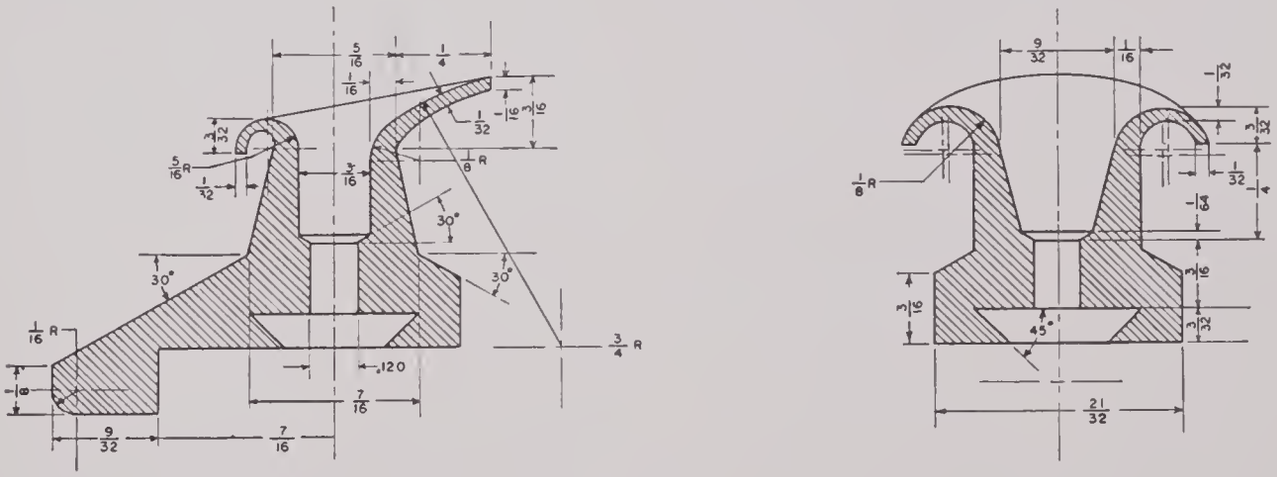


FIGURE 6. Diagram of semi-insert coupler, Harvard Design C-6. Vertical and horizontal sections.

11.2 HANDSETS AND EARPHONES FOR SPECIAL APPLICATIONS

In spite of the tremendous advances of radio communication in World War II, the field telephone, consisting of two handsets, batteries, ringers, and connecting line, still played an

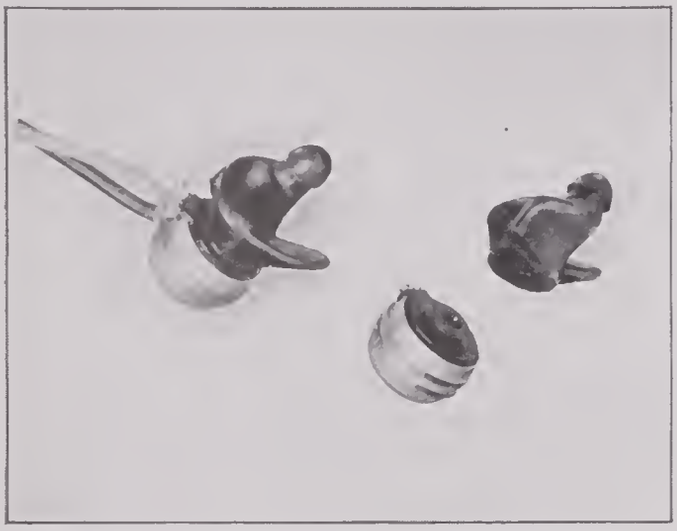


FIGURE 7. Harvintip insert tip, Harvard Design M-9.

extensive and indispensable role. In fundamental design, it was one of the best devices in the communication field. It had, however, one important defect. The handsets were of conventional design, similar to the standard telephone handset. As a result, they could not be used by the operator under the standard M-1

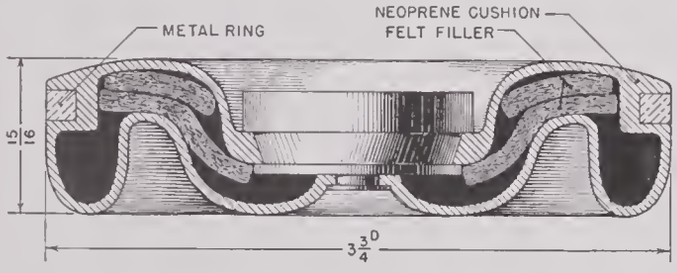


FIGURE 8. Cross section of dual-seal cushion.

steel helmet because of the bulky earphone. The helmet had to be cocked to the side, with a resultant decrease in protection (see Figure 10).

To overcome this difficulty, a handset of novel design was developed, based indirectly on a suggestion from captured German equipment. By utilization of the driving unit of the ANB-H-1 earphone, the earphone is considerably reduced in size, and the shape of the handset was redesigned for most convenient accommodation under the standard steel helmet. Figure 11 shows a subject using an experimental model (Harvard Type MTS-1) of this design under the helmet. Note that the presence of the handset does not necessitate a sidewise movement of the helmet. Figure 12 shows photographs of this handset taken from the front and side. An engineering drawing is shown in Figure 13.

In addition to the obvious advantages of this design over the conventional types used in the standard field telephone (Type EE-8), the Type MTS-1 handset has rubber cushions placed over the earphone which make for a better seal at the ear. The push-to-talk switch was completely

FIGURE 9. Photograph of dual-seal cushion.

redesigned. The design was adopted in substance by the U. S. Marine Corps.[7]

In order to provide privacy for the talker, which is necessary under certain conditions, a noise shield was added to the handset. A laboratory model of a handset which uses an ANB-M-C1 mask microphone in a noise shield is shown in Figure 14, a photograph of the instrument in use position; Figure 15 shows photographs of the instrument itself.[7a] Difficulty in satisfactorily fitting the wide range of face shapes and head sizes encountered led to the abandonment of the project.

The early engineering work on the "thin"

earphone used in the CW-49507A headset which has been adopted by the U. S. Marine Corps (see Figure 16) was done at the Electro-Acoustic Laboratory.[3] The magnetic structure of the ANB-H-1 earphone was redesigned to decrease the overall thickness at the expense of the diameter. The objective of accommodating the headset under the standard M-1 steel helmet was thereby largely achieved.

The enemy's policy of instructing snipers to concentrate on communication personnel suggested that some advantage might be gained by devising an earphone small enough to be concealed in the ear itself. Such an ultra-miniature earphone was under development by the Permoflux Corporation at the end of World War II. Approximately a centimeter long and a centimeter in diameter, the earphone itself is even smaller than the insert tip designed for it. When the wearer is viewed face-on, the earphone is not visible in his ear. A photograph of the experimental unit is shown in Figure 17.

FIGURE 10. Conventional field telephone. Handset cannot be used effectively under standard steel helmet.

FIGURE 11. Handset designed for use under standard steel helmet.

11.3 NOISE SHIELDS

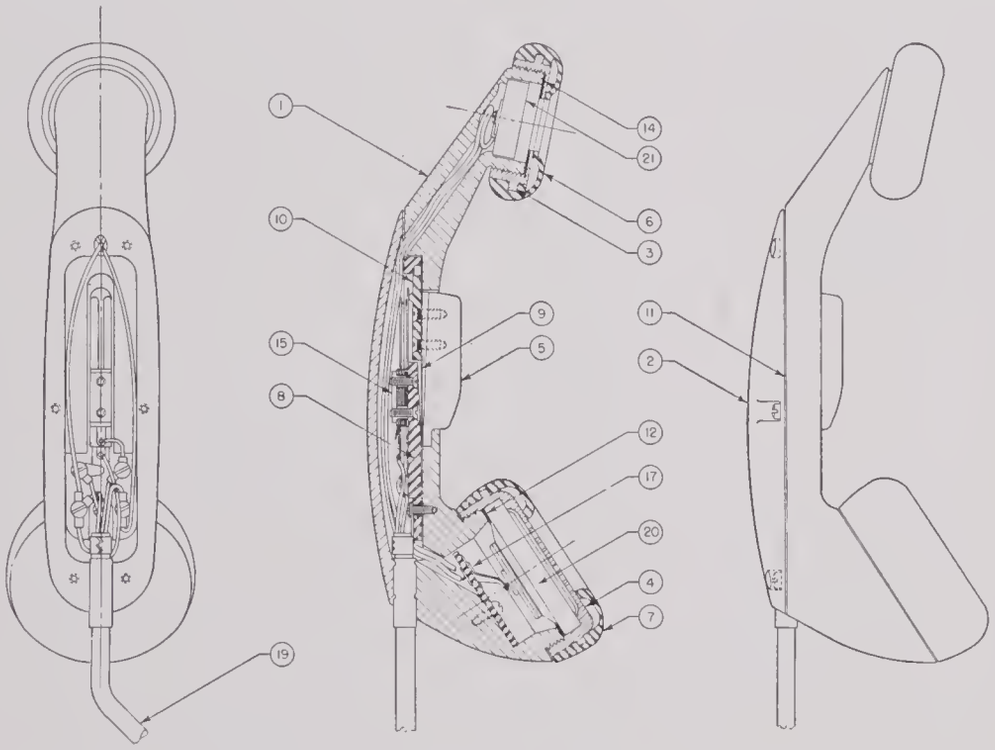
The most ambitious attempt to improve the discrimination between speech and noise at the microphone end was in the construction of noise shields which used actual mechanical shielding to exclude noise from the microphone. Several laboratories have tried their hands at finding a satisfactory design. At the Psycho-Acoustic Laboratory, 25 different models were tried at various times. The acoustical problems of noise exclusion and satisfactory frequency response were fairly well solved in the earlier models, but further effort was expended in an attempt to increase the comfort and to simplify the construction of molds for production purposes.

One of the earlier models, Harvard Design D-11 shown in Figure 18, indicates the main features which were incorporated. The noise shield holds an oxygen-mask microphone and provides a leak for the expired breath. It is

FIGURE 12. Photograph of model of improved handset.

held against the face by straps around the head. The cavity size was larger than necessary, and the face fit rather unsatisfactory. Both were rectified in the more conical shape of Harvard Design D-17, shown in Figure 25 of Chapter 9. Military personnel who used these models complained about having a device of this kind in contact with their faces, and the designers experienced difficulties in getting molds made properly. Although considerable interest was aroused in the D-17 design, it was never used for other than training purposes, and efforts to produce a noise shield were finally abandoned when the noise-canceling microphone appeared to meet the more pressing needs of the Air Forces.

Canadian interest in the noise shield resulted in the development of a model for use with a hand-held microphone. It was adopted for use with a dynamic microphone (see Figure 19). In the design of this shield an attempt was made to get a good seal against the face and yet retain a very simple conical construction. A diagram



ITEM	NAME	MATERIAL
1	HANDLE	ALUMINUM CASTING
2	REAR COVER	"
3	RECEIVER CAP	"
4	TRANSMITTER CAP	"
5	SWITCH BUTTON	PHENOLIC MOLDING
6	RECEIVER CUSHION	NEOPRENE GN
7	TRANSMITTER CUSHION	"
8	SWITCH MOUNTING	PHENOLIC
9	SWITCH GASKET (CEMENTED TO SWITCH MOUNTING)	BUNA "S"
10	SWITCH BUTTON STOP	PHENOLIC
11	REAR COVER GASKET (CEMENT TO HANDLE)	BUNA "S"
12	TRANSMITTER GASKET (CEMENT TO FACE OF TRANS. CAVITY)	NEOPRENE & BRASS
14	RECEIVER GASKET (CEMENTED TO FACE OF RECEIVER CAVITY)	"
15	SPRING & CONTACT ASS'BLY	—
17	CONTACT DISC ASSEMBLY	—
19	CORD ASSEMBLY	—
20	TRANSMITTER ASSEMBLY	—
21	RECEIVER ASSEMBLY	—

FIGURE 13. Drawing of improved handset.

FIGURE 14. Photograph of handset used with noise shield.

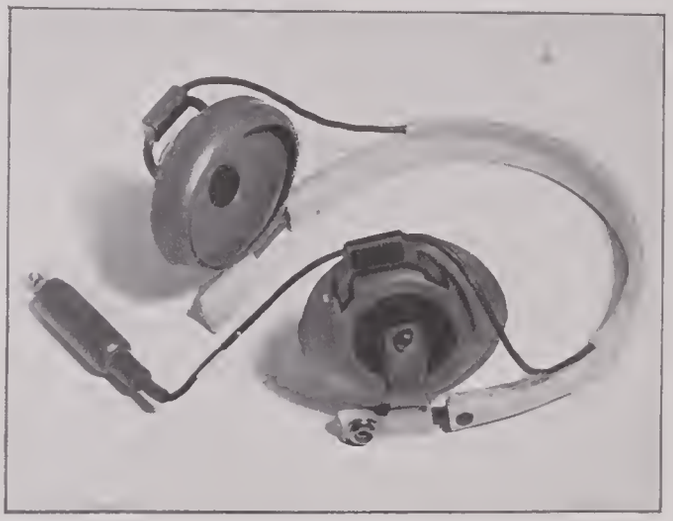


FIGURE 16. Photograph of headset assembly Navy Type CW-49507A (U. S. Marine Corps).

FIGURE 15. Photograph of model of handset with noise shield.

FIGURE 17. Photograph of ultra-miniature earphone with insert tip.

of the noise shield, Harvard Design D-25, is shown in Figure 20. The real-voice response of the Canadian unit is shown in Figure 21.

An air vent is, of course, a necessary feature of a noise shield and may reduce the mask's noise-excluding properties unless special care is taken in designing the opening. The Harvard designs successfully employed several rather long vents of small diameter. Such vents present little resistance to the direct flow of air but a high impedance to the passage of sound.

The optimum performance of a noise shield has probably not been obtained in any of the designs discussed above. The present limitation is almost certainly the severe acoustical reaction on the voice of the small cavity of the shield. Further improvements in the acoustical properties of noise shields will probably result from making the acoustic impedance lower and more resistive in nature. The inclusion of the nose under the mask, as in the Bell Telephone Laboratories design, has the advantage of including the nasal speech sounds. If the noise levels in aircraft continue to rise, it may eventually become necessary to adopt an improved shield as the only feasible method of obtaining the speech-to-noise ratio necessary for successful communication.

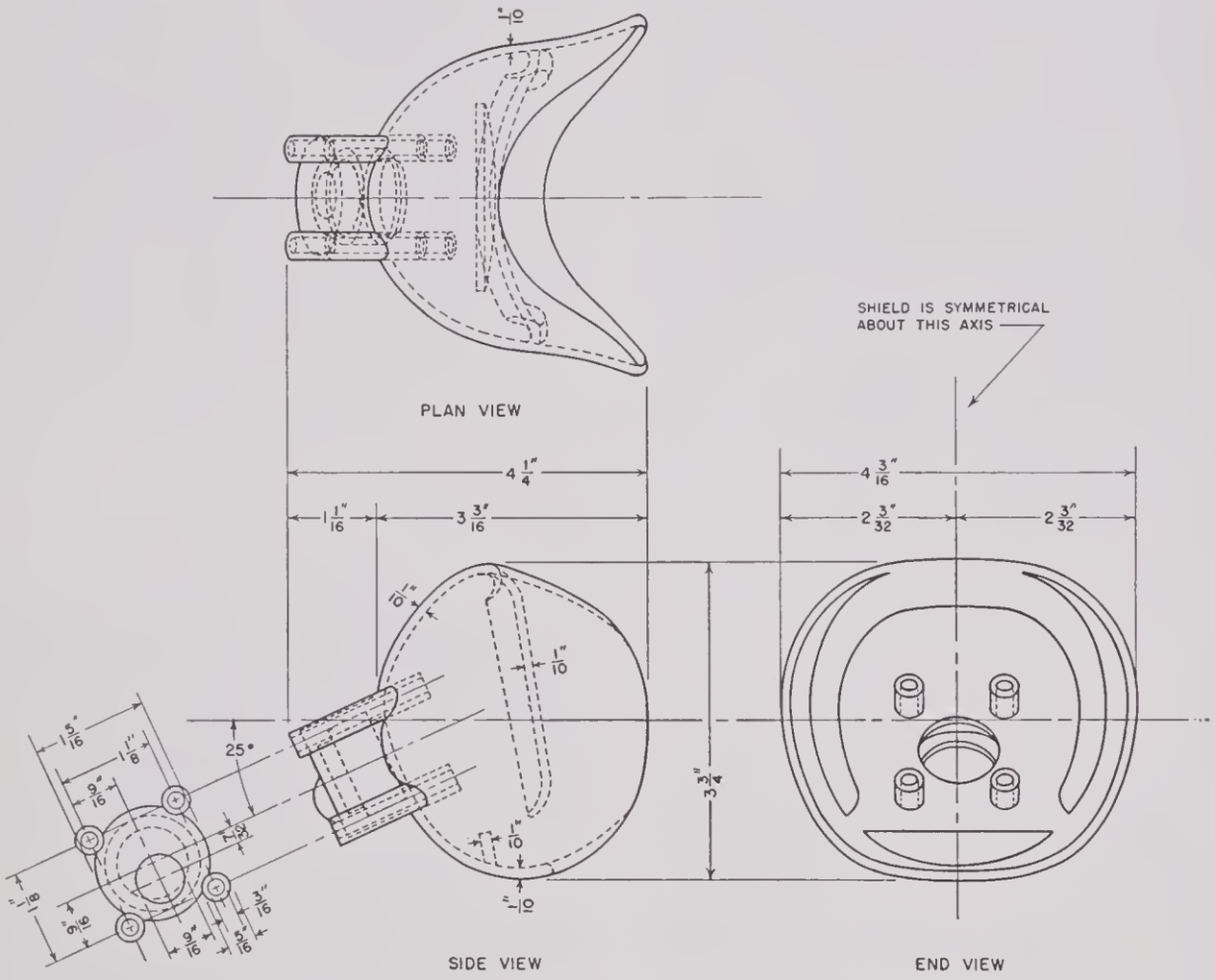


FIGURE 18. Drawing of noise shield, Harvard Design D-11.

11.4 OXYGEN MASKS

Some of the fundamental properties and characteristics of performance of oxygen masks have been discussed in Sections 9.2.4 and 10.2.1. In this section, certain tests made on an artificial voice are discussed which may be helpful in design problems involving oxygen masks. At the outset, it must be emphasized that the fundamentals of the performance of enclosures coupled to the mouth are, as yet, imperfectly

FIGURE 19. Canadian hand-held dynamic microphone with Harvard Design D-25 noise shield.

understood. Many of the results must be interpreted with this fact in mind.

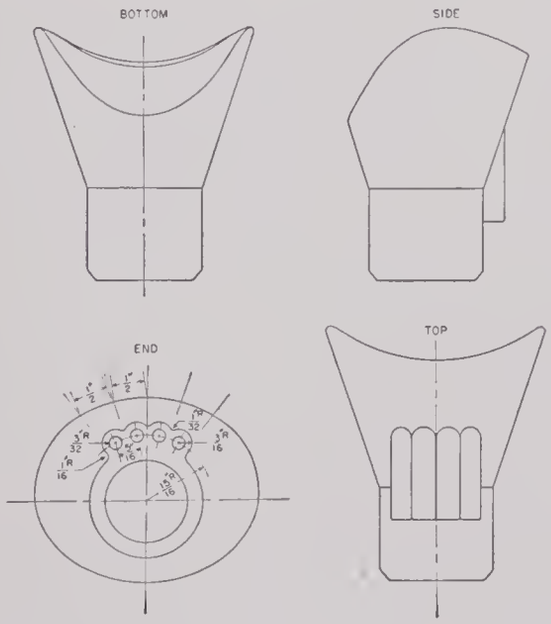


FIGURE 20. Drawing of noise shield, Harvard Design D-25.

Early experiments with small enclosures coupled to an acoustic source of high internal impedance[2] have demonstrated two things. First, the response of a mask microphone per se must have a rising frequency-response characteristic at low and medium frequencies in order that a satisfactory response characteristic of the mask-microphone combination be realized. Second, the distribution of sound pressure in the mask becomes nonuniform at medium and high frequencies and proper microphone location within the mask is important.

In addition, many experiments have been carried out in order to determine the effects of the oxygen supply hose, mask size and seal to the face, the noise exclusion properties of oxygen masks, and similar factors. Some of the relevant data are presented in report form.[5]

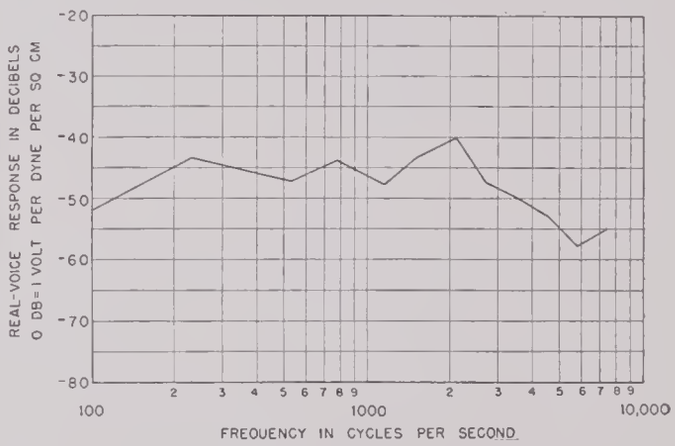


FIGURE 21. Real-voice response of Canadian ZA/CAN 5155 hand-held microphone with Harvard Design D-25 noise shield.

There remains a pressing need for consideration of acoustical factors at an early stage in the development of any future masks.

While the artificial voice is a basic tool in the hands of the designer, its chief merit lies in its simplicity and the speed with which *comparative* results can be obtained. Real-voice response measurements are an important supplement and should also be utilized. But certain acoustical properties of oxygen masks and noise shields, as yet improperly understood, cause speech produced within them to be seriously distorted. In many cases, significant differences in articulation scores exist between masks which real-voice response characteristics fail to predict. Formal articulation tests remain, therefore, the only method which will give a high correlation with actual performance.

11.5 GAS MASKS

At the request of the U. S. Army Chemical Warfare Service, an investigation of the acoustical properties of gas masks was undertaken by the Harvard group. These properties are important in two related ways. First, it is necessary for combat personnel exposed to gas to communicate with each other directly; second, successful speech transmission over field telephones or over battle telephones on shipboard is necessary.

FIGURE 22. Gas mask without diaphragm.

FIGURE 23. Gas mask with diaphragm for improved speech transmission.

In the masks intended for use at the beginning of World War II, the exhaust valve was the chief path of acoustic transmission to the outside. The acoustic ineffectiveness of these types soon became apparent.[4] Masks of later design were equipped with a diaphragm (speech transmitter) which was frequently combined with the exhaust valve (angle tube). Figure 22 shows an early mask without speech transmitter. A mask which utilizes an angle tube (combination diaphragm and exhaust valve) and which was proposed for use early in 1944 is shown in Figure 23.

Articulation tests[4] have shown that masks equipped with diaphragms are acoustically superior to masks relying on the exhaust valve alone for speech transmission. However, the articulation scores for *all* masks tested were substantially lower than the scores obtained for talkers without masks. This was found to

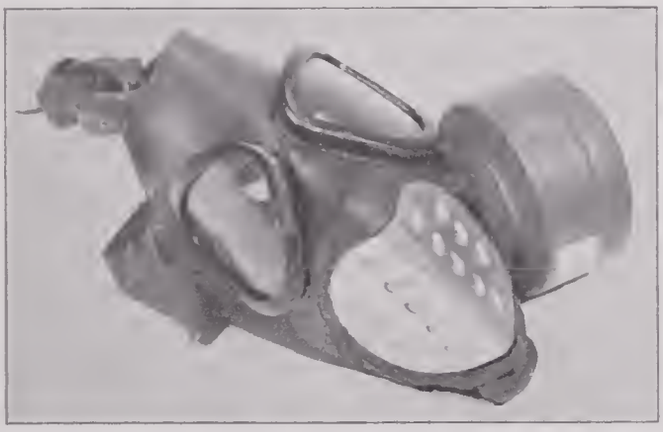


FIGURE 24. Photograph of laboratory model of gas mask with improved speech transmitter.

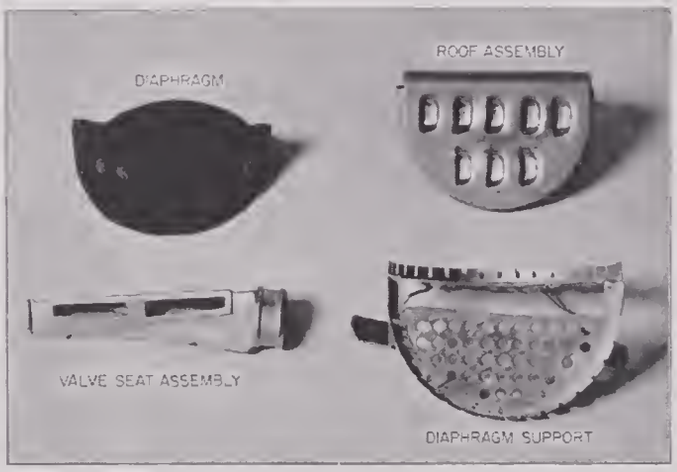


FIGURE 25. Photograph of speech transmitter parts (Harvard Design).

be true for articulation tests conducted in a low ambient noise as well as for tests conducted outdoors.

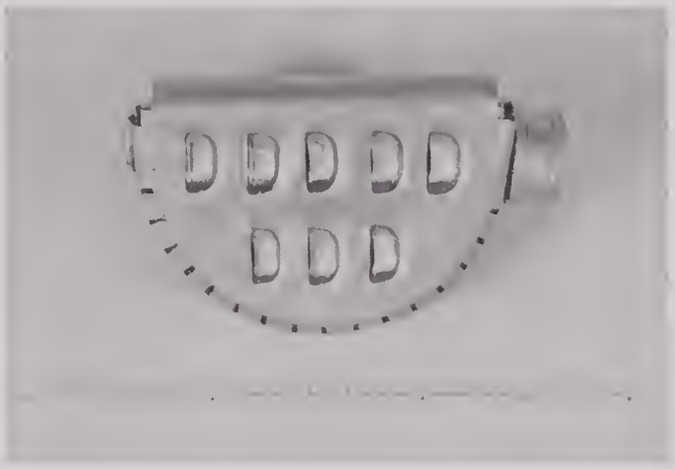


FIGURE 26. Photograph of speech transmitter assembled (Harvard Design).

In articulation tests where the person wearing a gas mask tried to use the standard field telephone, the observation was made that speech transmission was intolerably poor unless the microphone of the handset was held very closely to the exhaust valve or diaphragm. Since this necessitated the removal of the earphone from the ear, it became clear that a complete redesign of mask (rather than of the telephone) was in order. This was equally evident after a series of experiments attempted to improve speech transmission into open air by minor modifications of the diaphragm mask.

Physical tests were performed using real and artificial voices,[4a] but it soon became evident that these tests did not tell the whole story, since interaction between the speech mechanism and the mask was an important, if not controlling, factor. The phenomena appear even more complicated than those present in noise shields and oxygen masks, since gas masks involve sound transmission through the mask to the outside.

During 1945, an attempt was made to design a new speech diaphragm which, in conjunction with the mask, would provide more effective speech transmission into open air and over standard field telephones.

Figure 24 shows a laboratory model of the

FIGURE 27. Photograph of boom mounting of lip microphone in use position (front view).

FIGURE 28. Photograph of boom mounting of lip microphone in use position (side view).

design which was finally evolved. It shows the speech transmitter (E17R8) mounted on a facepiece with lateral canister.[8] Figure 25 shows the parts of the speech transmitter. The active part of the valve is a rubber strip of V-shaped cross section, with semicircular diaphragms extending from it. The valve rests against the valve seat, whereas the diaphragm is constrained by the metal diaphragm support. The

FIGURE 29. Photograph of boom mounting of lip microphone in standby position.

structure is protected from the outside by a metal roof. The assembly is held together by a crimped edge (see Figure 26). Air pressure from the inside of the mask opens the valve. The perforated metal diaphragm support furnishes effective protection against gun blast, whereas the roof provides protection against diaphragm puncture and foreign matter.

Articulation tests have shown that a mask equipped with the new E17R8 speech transmitter is somewhat superior to older diaphragm masks. Performance over field telephones should be markedly better, since it is no longer necessary to remove the handset from the ear in order to place the microphone against the opening in the mask. But despite the fact that a significant improvement has been realized, the basic problems have not been solved.

An attempt to attack the problem of interaction between mask and voice in a thorough way has been made.[8a] The effects of small, idealized enclosures terminated in openings of various sizes has been studied. An artificial voice was used, similar to the one discussed in Chapter 10 (shown in Figure 28 of that chapter). It was transformed effectively into a constant current source by packing the tube simulating the mouth with wires. The sound pressure developed in the cavity was measured by means of a probe microphone and was taken to be proportional to the acoustic impedance of the enclosure. By comparing the sound pressures inside and outside the cavities, an index of the transmission of sound was obtained. The effect of such enclosures on the live voice was also studied, and many interesting results have been obtained.[8b] Much research of a fundamental nature, however, remains to be done before the mechanism of interaction between voice and mask is understood.

11.6 BOOM MOUNTING FOR MICROPHONES

The advantages and favorable performance data of the "lip" microphones of the noise-canceling type have already been pointed out (see Section 9.2.2). A photograph of the Navy noise-canceling microphone M-5/UR with harness is shown in Figure 16 in Chapter 9. As a result of field tests, certain objections to this type of suspension were raised, a summary of which follows.

1. The harness, in direct contact with the lips, is uncomfortable and might cause skin irritation, especially in the tropics.

2. The harness sometimes causes pressure of the microphone against the septum of the nose, a further cause of discomfort.

3. No provision is made for a convenient standby position.

4. Large accelerations in pull-outs tend to displace the microphone down over the mouth.

To correct this situation, the U. S. Navy Bureau of Aeronautics encouraged the development of an alternative mounting. A solution was found in the form of a boom mounting. The microphone is supported in a boom made of stiff wire which is supported on a pivot on the earphone cushion. Photographs of this mounting are shown in Figures 27 and 28. The boom in standby position is shown in Figure 29. Various orientations of the microphone button with respect to the boom were tried and the performance was measured in terms of articulation score and noise pickup (see Section 10.1.2).[7b] These mountings were available for use by the U. S. Navy at the end of World War II.

Chapter 12

SPECIAL VOICE COMMUNICATION SYSTEMS

12.1 SOUND-POWERED TELEPHONE SYSTEMS

A SOUND-POWERED telephone system is an interphone system (see Figure 3, Chapter 9) *without* an amplifier or any other source of

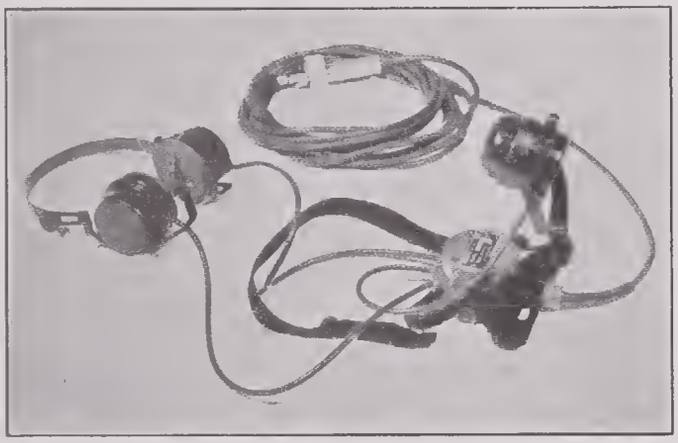


FIGURE 1. Photograph of Type M sound-powered telephone instruments (U. S. Navy).

electric energy. The energy necessary to operate the system is supplied solely by the talker's voice. The separate lines necessary in a con-

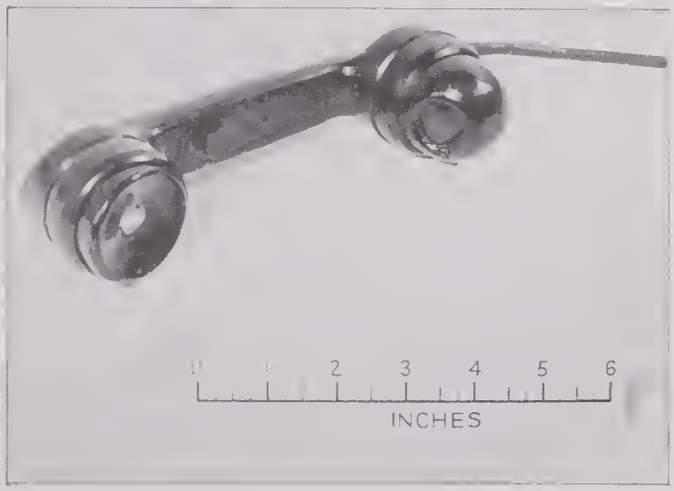


FIGURE 2. Photograph of sound-powered handset (U. S. Navy).

ventional interphone for headsets and microphones are combined, and the network connecting the stations is a simple two-wire line.

Sound-powered telephones are used most extensively aboard ships. A great number of circuits are provided, and the number of stations per circuit is generally large, sometimes exceeding twenty-five. Sound-powered handsets have also been used, to some extent, as field telephones by the Ground Forces.

On shipboard the bulk of the interior communications during General Quarters is carried by sound-powered telephones. The operators manning the stations consist of enlisted personnel equipped with headsets and microphones mounted on chest plates. Figure 1 shows an instrument of this type (Type M) using earphone cushions with a supra-aural seal. These sound-powered telephones were in use through-

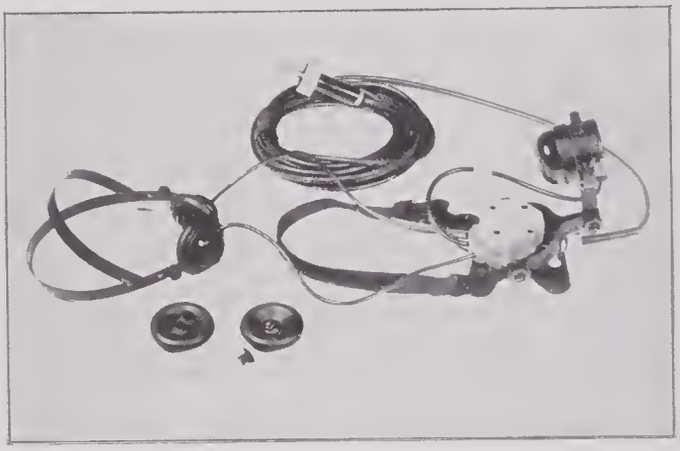


FIGURE 3. Photograph of Type O sound-powered telephone instruments (U. S. Navy).

out World War II. Certain key officer personnel below deck, for example, make use of sound-powered instruments combined in a handset (see Figure 2).

At the close of the war, headsets of Type M were being replaced by units of decreased size and weight (see Figure 3) utilizing removable semi-insert tips (Type O). This change was intended to overcome some of the criticisms leveled against the older headsets, such as their excessive weight and bulk, the necessity of using a special large helmet at exposed stations, and the discomfort which resulted from prolonged wear. While these objectives were achieved in some measure, the performance of Type O headsets fell below that of the Type M in certain other important respects. Extensive

articulation tests in noise showed that the Type O units were comparable to Type M units only if used with the tips, and then only if great care was taken to obtain as good a seal

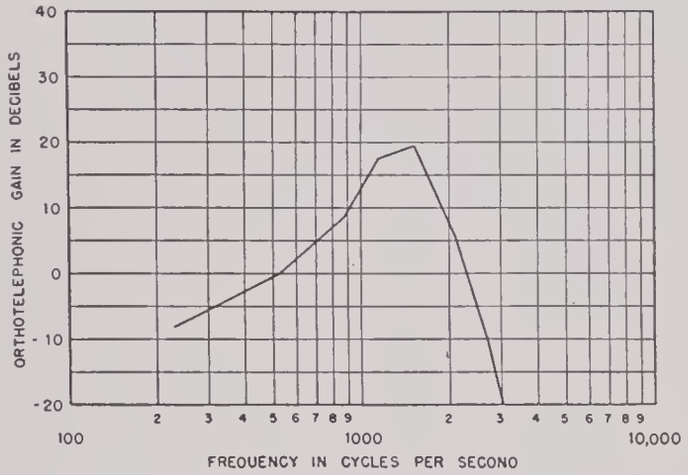


FIGURE 4. Orthotelephonic response of a typical sound-powered telephone system (Type M).

on the ear as possible. The performance of the Type O units, used *without* the tips, was shown to be definitely inferior.[4, 5, 9] Reports from the Fleet indicate that there is little likelihood that the tips will be properly used or used at all by combat personnel.

Further performance data (response and insulation) of Type O headsets are given elsewhere.[7, 12] It is understood that the U. S. Navy Bureau of Ships is now procuring instruments of novel design which may overcome some of the present difficulties.

In the course of testing the many particular designs used or proposed for use during World War II, quantities of more general information have been gathered. The remainder of this section is devoted to a discussion of some of the general properties of sound-powered telephone systems and their performance in noise. For simplicity, Type M units will be assumed throughout the discussion.

The efficiency of sound-powered earphones and microphones in transducing sound into electric energy and in again converting the electric energy back into sound must be made high to compensate, if possible, for the lack of external amplification. This has been achieved to a considerable degree in practical designs but only at the expense of the *bandwidth* of the system. Figure 4 shows the overall orthotelephonic gain of a typical sound-powered telephone system. It can be seen that this system transmits efficiently only a relatively narrow band of speech, about one octave wide, centered about a frequency of 1,500 c. The maximum orthotelephonic gain, near 1,500 c, is about 20 db, i.e., the level of received speech is 20 db higher than it would be over an air path of 1 m.

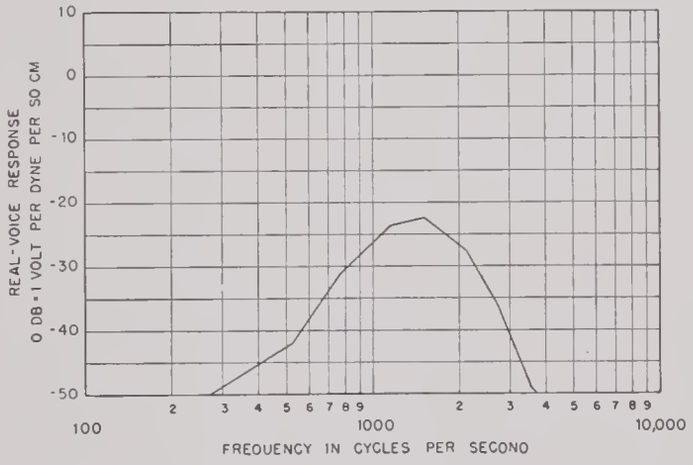


FIGURE 5. Real-voice response of a typical sound-powered microphone (Type M).

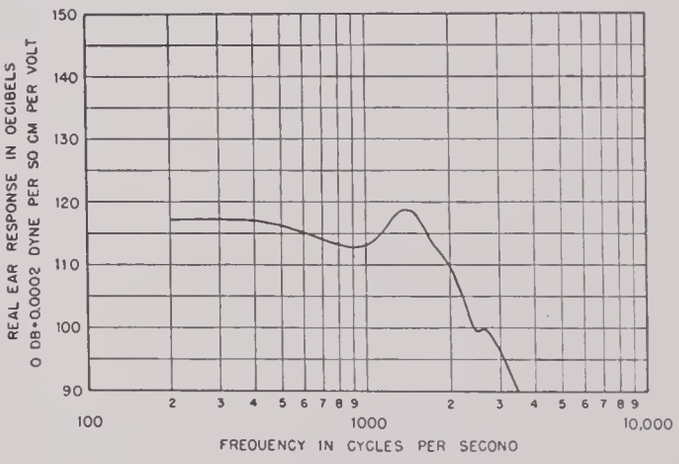


FIGURE 6. Real-ear response of a typical sound-powered earphone (Type M).

The rapid decrease in the response above 1,500 c is due to both earphone and microphone (see Figures 5 and 6). The decrease in the response at the lower frequencies is due mainly to the microphone.

It is evident at once from Figure 4 that the effective pass band of such a system, i.e., the band of frequencies where the speech-to-noise ratio is contributing significantly to the articu-

lation index (see Chapter 9), is a function of the noise level, since there is a quite pronounced upper limit to the speech level a talker can maintain comfortably. In noise, a band of frequencies even less than an octave wide may be effective, whereas in quiet surroundings the effective band will exceed two octaves. If articulation tests in varying levels of noise are performed on such a system, it is possible to "bracket" the resulting function by corresponding gain functions (see Chapter 5) of two systems with sharp cutoffs and suitable band-

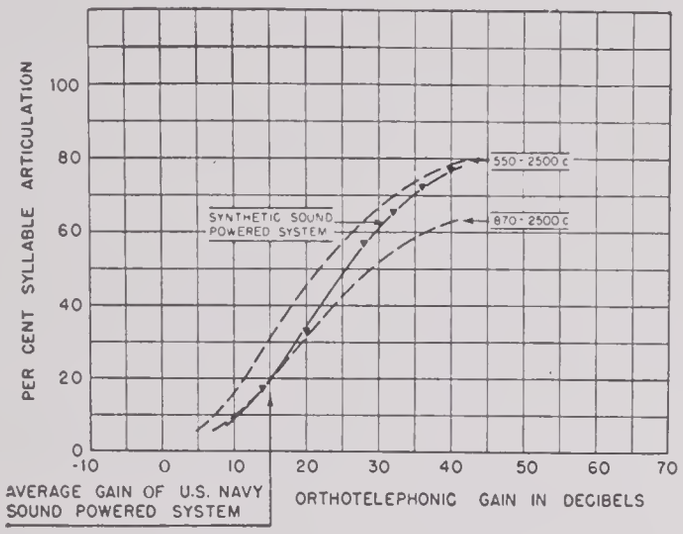


FIGURE 7. Gain function obtained with a synthetic sound-powered system in noise. For comparison, gain functions for two band-pass systems with sharp cutoffs are shown also.

widths. Such tests have been performed using a synthetic sound-powered system closely approximating the real one. A noise spectrum was used resembling the spectrum in a submarine engine room. Then, with the talker located in a quiet place and speaking in a loud voice, the articulation functions shown in Figure 7 were obtained. Although, for these tests, the gain of the synthetic system was varied by means of an amplifier, essentially the same functions would have been obtained if the noise levels had been changed in an inverse propor-

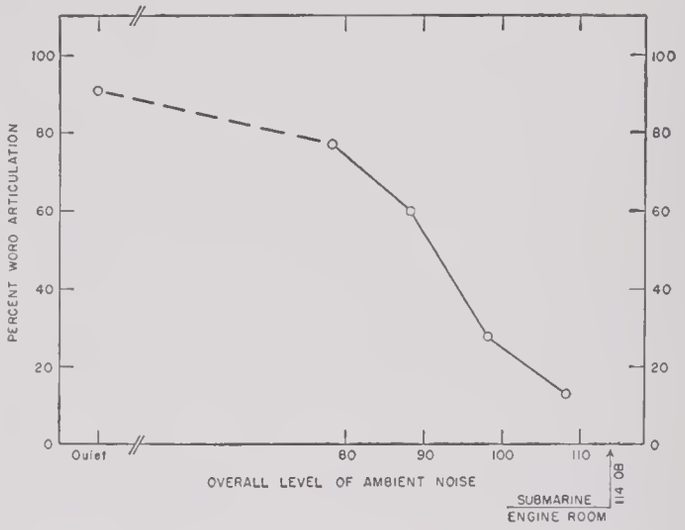


FIGURE 8. Showing the performance of a typical sound-powered telephone system in ambient noise.

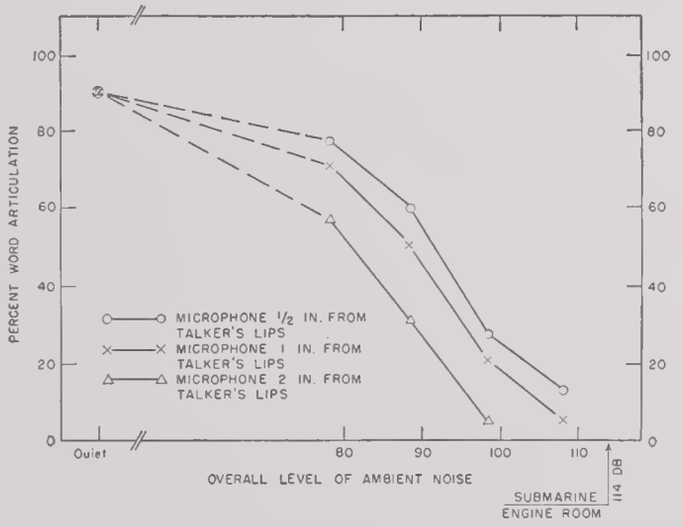


FIGURE 9. Showing the influence of the spacing between the microphone and the talker's lips for a typical sound-powered telephone system.

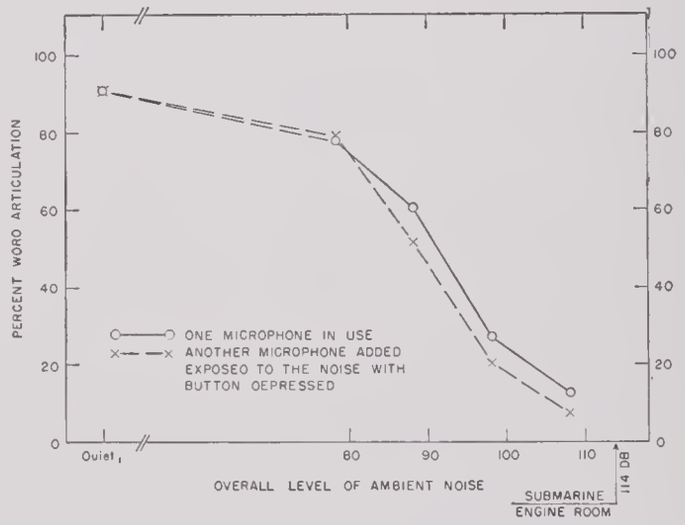


FIGURE 10. Showing the influence of an "open" microphone on the performance of a typical sound-powered telephone system.

tion while an amplifier of fixed gain (zero) was being used. The functions for the two "bracketing" systems with sharp cutoffs are also shown.

From these functions, it can be concluded

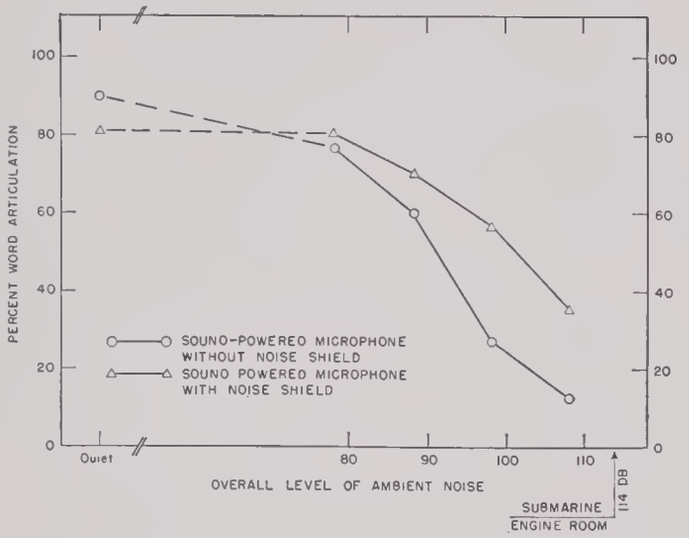


FIGURE 11. Showing the improvement in articulation score afforded by the use of a noise shield over the sound-powered microphone.

that the performance of a typical sound-powered telephone system is adequate in relative quiet only. Under some noise conditions encountered on shipboard the performance

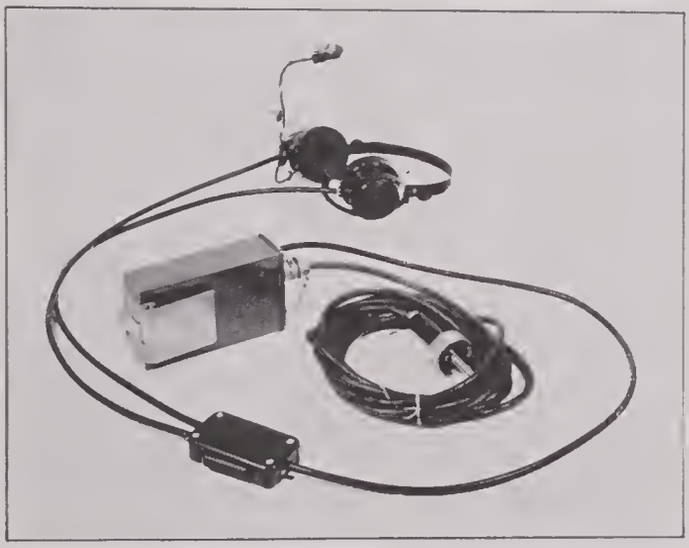


FIGURE 12. Noise-canceling carbon microphone Type M-6/UR with adapter kit for use on a sound-powered telephone line.

breaks down. This has been confirmed by analyses of recordings taken from actual sound-powered circuits while they were in normal use.[3] A somewhat more precise demonstration of the effect of noise in limiting actual performance is afforded by laboratory articulation tests. Figure 8 shows the articulation scores obtained when both listener and talker are immersed in "engine-room" noise. At the level actually found, for example, in a submarine engine room, the low articulation score indicates that the system will fail seriously because of the many necessary repeats.

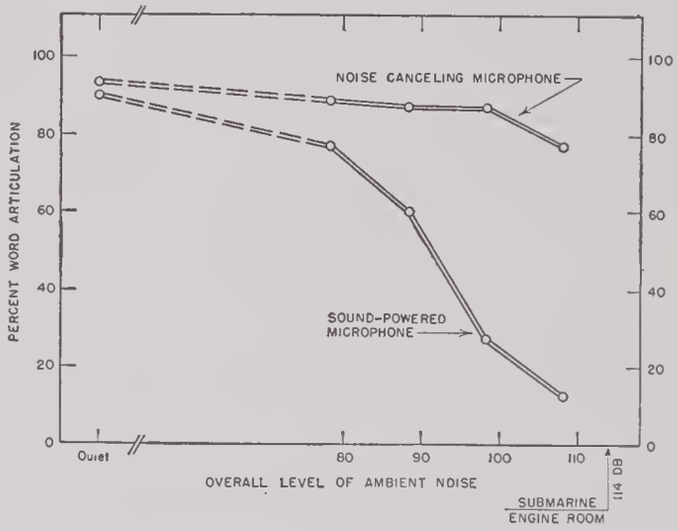


FIGURE 13. Showing the improvement in articulation scores obtained by using a noise-canceling microphone in a sound-powered telephone system.

With the talker alternately in quiet and in the same noise as the listener, articulation scores obtained on a sound-powered system in-

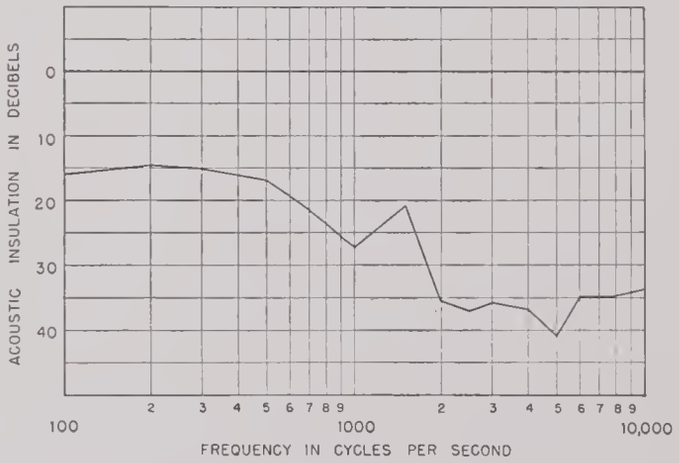


FIGURE 14. Acoustic insulation provided by sound-powered earphone (Type M).

dicate that the noise picked up by the microphone contributes significantly to the masked threshold in the pass band. This is true even for close-talking conditions. As the number of headsets is increased, the relative contribution

by the microphone to the total masking decreases, and, for a very large number of headsets across the line, the noise leaking past the earphone cushion is the controlling factor.

The sound-powered telephone system performs satisfactorily under conditions of relative quiet. Furthermore, such a system is the simplest one possible. Its reliability is high if it is properly designed. It is less vulnerable than a system with central batteries, although the present-day practice of terminating many lines in the switching facilities of the Internal Communications Room aboard ship makes it more vulnerable to a "lucky" hit than would appear at first glance.

On the other hand, there is ample evidence that the sound-powered system fails under the stress of ambient noise. This is due not only to the properties of the system per se, as shown above, but can also be attributed in no small measure to the present doctrine of using telephone "talkers." More often than not, the telephone line terminates in a "talker" rather than the man who is charged with executing the transmitted order. The talker's function is solely that of a relay, transmitting and receiving messages in open air to and from a second person who receives or gives the order. This procedure may be necessary in many instances; it is as often unnecessary and ineffective. The talker is often the weak link in the system, for in noise he has difficulty in hearing the spoken order while he is wearing the sound-powered headset. Correspondingly, it is difficult for the talker to shout the message through the open air. Detailed studies have shown that on some circuits, mainly those concerned with fire control, the number of open-air links is far too large for satisfactory performance.

Adequate training and correct attitudes of personnel are important. Good enunciation, high speech levels, and use of appropriate speech material are helpful. The microphone should be held as closely as practicable to the talker's lips. Figure 9, for instance, shows the drop in articulation scores as the distance between microphone and the talker's lips is increased. The microphone press-to-talk switch should be operated only if a message is to be transmitted. One "open" microphone on the line has the effect shown in Figure 10. All headsets connected to the line should be worn with as good a seal as possible at all times.

It seems probable that a proper study of the systems and circuits and coordination of this study with improved doctrine would outline improvements which would raise the operating efficiency of intraship circuits. Then, as a further step, the possibilities of improving the sound-powered system proper should be investigated.

The signal-to-noise ratio provided by present-day sound-powered microphones can be improved by the use of a noise shield. A laboratory model of such a shield was produced by modification of the Harvard D-17 noise shield to fit the sound-powered microphone (see Section 11.3). Figure 11 shows the improvement in articulation scores resulting from its use.

As a further step, use might be made of carbon microphones of the noise-canceling type. An adapter kit[11] (see Figure 12) was devised utilizing a Type M-6 UR lip microphone mounted on a boom suspended from the earphone. The kit includes the necessary dry cells, transformer, and switch. The improvement in articulation score resulting from the use of the adapter is quite striking (see Figure 13).

Tests show that the Type M headset excludes noise quite well. To achieve this, however, the headband exerts a large pressure on the ears, and the earphones are of considerable weight (see Figure 14). The sensitivity compares favorably with other types of headset. The frequency responses of sound-powered microphones and earphones (see Figures 5 and 6) show that, at present, the bandwidth of the system is largely limited by the microphone. It seems unlikely that the efficiency can be further increased without narrowing the pass band beyond tolerable limits. Manufacturing tolerances, available magnetic materials, impedance mismatch between microphone and earphone, and rigid specifications for resistance against corrosion and blast set severe requirements for practical designs.

The question arises, therefore, whether present-day response curves constitute the optimum consistent with current design requirements. Suppose a system were built with

a pass band one octave wide but centered around 1,000 c instead of 1,500 c, the orthotelephonic gain being the same in both cases. How would the performance of these two systems compare? Such questions can be answered in general by making use of the generalized equal-articulation contours (see Section 7.2.4) obtained from a thorough study of the articulation efficiency provided by bands of speech in noise. From the more detailed discussion presented in an earlier report,[6] it can be concluded that no net gain in performance will result from the shift in frequency mentioned above. A significant improvement will be obtained only if the orthotelephonic gain or the bandwidth or both are increased.

With these severe limitations imposed on a sound-powered system, the question arises whether it may not be more profitable to abandon the sound-powered principle for the battle telephone, or at least relegate it to its original standby status, and install an interphone system using efficient wide-band instruments and electronic amplifiers. The idea of using a central amplifier for each circuit, similar to the interphone system used in aircraft, seems attractive. On the other hand, individual station-amplifiers could be built. The problem is a difficult one to solve in practice, especially when questions of doctrine, procurement, and investment in existing facilities must be balanced against the desired performance. It is hoped that a thorough survey will be undertaken to re-evaluate the present system of internal communications over sound-powered telephones, and that this, supplemented by suitable research, will result in recommendations for future designs.

12.2 PUBLIC-ADDRESS SYSTEMS

Sound systems of the public-address type have been used for a variety of purposes in World War II. On shipboard, battle-announcing systems are in use to transmit general announcements and orders, calls to battle stations, and certain special code signals. On carriers, instructions and orders are transmitted to the hangar and flight decks.

On the ground, sound systems are used to direct certain battle and landing operations and transmit propaganda to nearby enemy installations. Some of the systems used by the U. S. Marine Corps have been tested.[10] The more important performance parameters are the maximum undistorted acoustic output, the frequency-response characteristic, directivity, and nonlinear distortion.

12.3 PORTABLE RADIO EQUIPMENT

The overall response characteristic of a communication system is obviously the cumulative result of the response characteristics of its components. If several of the components pass only relatively narrow bands of frequencies, the total effect is sometimes striking. The performance is sometimes determined by the single component with the narrowest pass band. A case in point is the sound-powered telephone system. Another instance was provided by portable radio equipment. This was brought to the attention of the Harvard group in 1943.[1] The system in question, used extensively in World War II, is shown in Figure 15. Upon investigation, it turned out that the earphone and microphone were the limiting elements in determining the quite unsatisfactory overall frequency response. There is evidence that, as is too often the case, the designers of the various equipments went their own way without much regard for coordination and overall performance. By substituting more suitable earphones and microphones, the overall frequency response was improved, as shown in Figure 16.

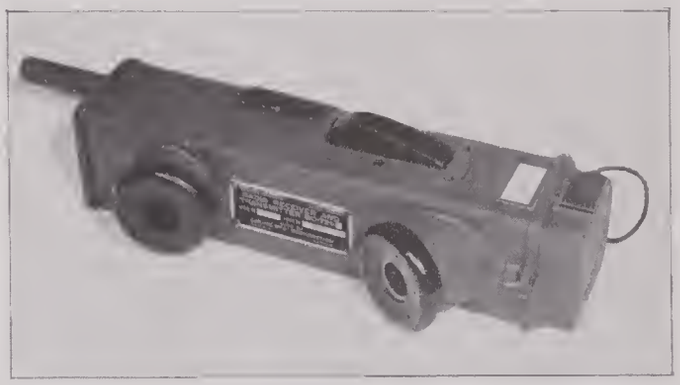


FIGURE 15. Portable radio transmitter-receiver.

12.4 COMMUNICATION SYSTEM FOR SHALLOW-WATER DIVING

The recent development of improved and simplified equipment for shallow-water diving led the U. S. Navy Bureau of Ships to request assistance in the design of a communication system suitable for use in improved diving dress. The system which was developed[8] enables two divers, operating in shallow water, to communicate with each other and with the tender. A set of earphones and a microphone are provided for each of the three operators, and the three are linked together by a portable, battery-operated amplifier and suitable cables.

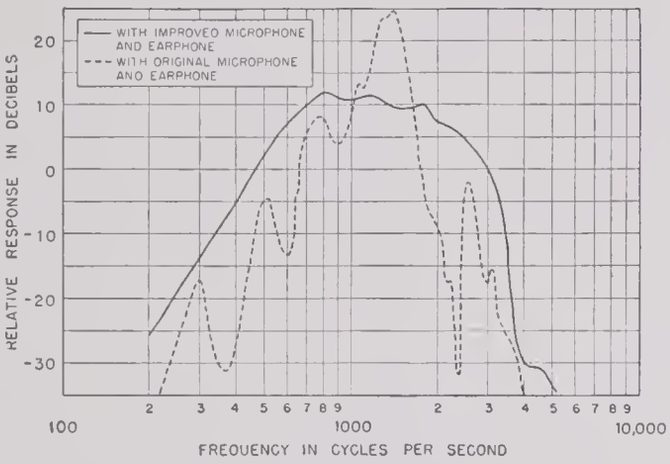


FIGURE 16. Improvement in overall frequency response of the radio transmitter-receiver by the use of a more suitable earphone and microphone.

Photographs of the diving suit before installment of the headset and microphone are shown in Figure 17. It consists of a watertight diving dress which covers all but the diver's face. A mask is strapped over the diver's face with a triangular plastic window in front. An intake control valve is provided for air which is supplied from a tank on the tender. The suit was modified to accommodate in the proper position two earphones sheathed in rubber. A special socket was devised to house an earphone unit similar to the one used in the Marine headset CW-49507. The leads are of a special type and are provided with a special air duct to achieve pressure equalization between the ears and the mask cavity. All junction points are carefully made watertight.

FIGURE 17. Photograph of a diving suit designed for operation in shallow water. No communication equipment is as yet installed.

The microphone used in the mask is of the noise-canceling type (CW-51066). It is mounted

on an adjustable bracket. Although the microphone was originally designed for operation in open air, its use in an enclosure does not result in an increase in the low-frequency response, as would be the case for ordinary microphones. This is due to the fact that, at low frequencies, the pressure gradient in the mask is comparable to that in open air, so that the response of the lip microphone in the mask is not greatly increased over what it would be in open air.[a] It was found necessary to reduce the noise generated by the air flow to a tolerable magnitude by installing a suitable silencer. A sketch of the diving suit and mask fitted with the communication instruments is shown in Figure 18. The tender uses ANB-H-1A earphones and a noise-canceling microphone with boom suspension (see Figure 19).

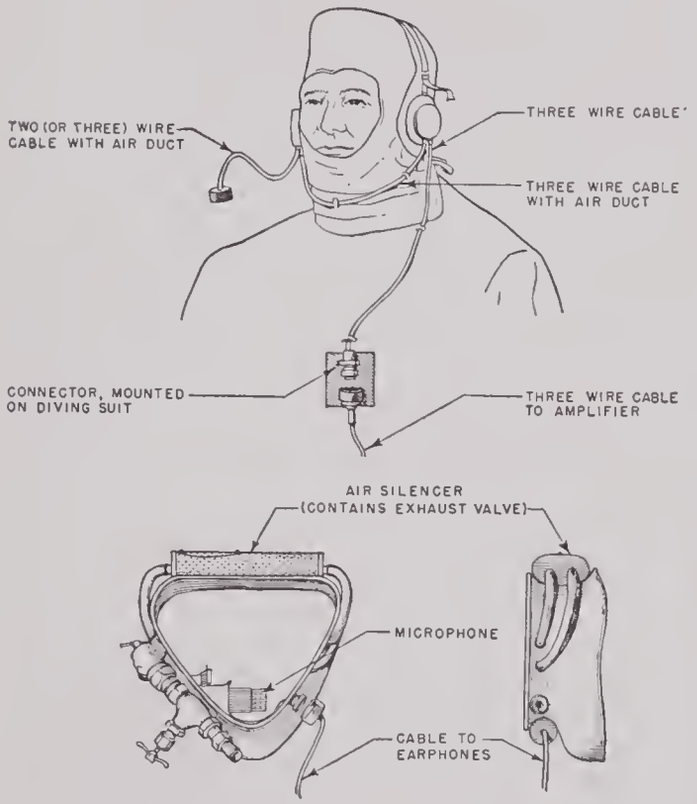


FIGURE 18. Sketch of diving suit and mask fitted with earphones and microphone.

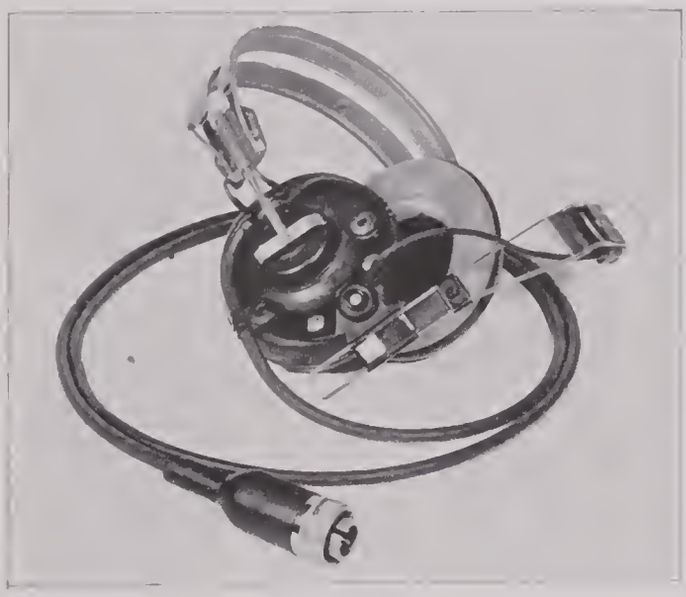


FIGURE 19. Photograph of headset and microphone used by tender.

A photograph of the amplifier is shown in Figure 20. It is battery-operated and contains carbon amplifiers and the necessary switches and cable connections. Its weight is approximately 25 lb. The overall frequency response of the system is essentially uniform to about 3,000 c. The change in response resulting from the increased ambient pressure at a depth of 60 ft has been measured and found to be not significant.

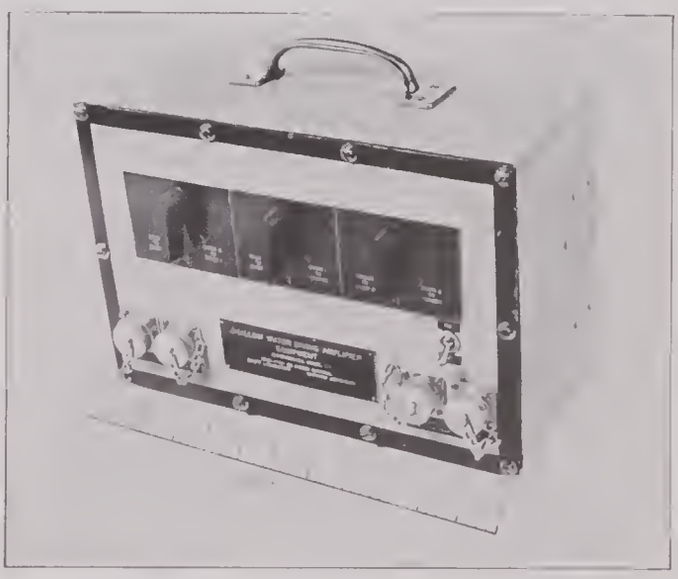


FIGURE 20. Carbon amplifier for communication with two divers operating in shallow water.

Figure 21 shows an early laboratory model of the modified diving dress with a junction box atop the diver's head. This design was later abandoned in favor of the arrangement sketched in Figure 18.

[a] Noise-canceling lip microphones have also been used successfully in oxygen masks.

FIGURE 21. Showing experimental installation of earphones in the diver's suit. (The junction box was later abandoned. See Figure 18.)

A laboratory model of the equipment has been repeatedly tested in a pressure tank and has met with general approval by the Navy.

12.5 VOICE TUBES IN SUBMARINES

In spite of all the advances made in the communication art, the voice tube has remained the unchanging standby system for use when all others fail. In some instances, such as between the bridge and conning tower of submarines, communication proved to be very ineffective. After extensive tests had been conducted, it was shown that proper design of the tubes and junctions, the presence and design of valves in the voice-tube circuit, and appropriate choice of horns at the talking and listening ends are all of critical importance. The submarine differs from other ships only in the severity of the space requirements, leading to improper location of lines, and number of valves required on lines passing through the hull. Full results and recommendations are presented elsewhere in report form.[2]

Chapter 13

NOISE REDUCTION IN RADIO RECEIVERS

THE RADIO LINK, like the interphone, is often the victim of interfering noise, and satisfactory speech communication depends in large measure upon the establishment of a favorable signal-to-noise ratio. Atmospheric static is usually the greatest problem to be overcome, although ambient noise, precipitation static, interference from radar modulators and electronic altimeters, ignition noise, interference from electric motors, interchannel interference or deliberate jamming, and set noise from the receiver itself are all too often encountered.

13.1 STATIC

A large part of atmospheric static, like most of the specific interferences mentioned, belongs to a class of interference which may be referred to as impulse noise. The intense static from local thunderstorms, and probably most other types of atmospheric static, arise from very brief electric or electromagnetic disturbances. These disturbances frequently occur in bursts, trains, or crashes, and the times of occurrence of the individual static pulses are, if not completely random, at least highly irregular. The individual impulses are also irregular in amplitude. The overall spectrum of atmospheric static varies from day to day and from season to season, but static is usually most intense through the low and medium ranges of radio frequencies. The spectrum extends, however, into the high-frequency range and occasionally even beyond. Within the pass band of a radio receiver, which is very narrow relative to the overall range of the static spectrum, the average distribution of static energy is essentially uniform.

As a communication hazard, the most important characteristics of a particular sample of atmospheric static are its *intensity level* and its *density*. Density is simply a matter of pulse repetition frequency [PRF]. Although for a particular sample of static the number of pulses per second may vary within relatively wide limits, it is possible to think of the static as having an average PRF. For various samples, this average PRF may range from essentially zero to exceedingly high values, depending upon weather and propagation conditions. When the PRF is low and the pulses are widely spaced, the intermittent character of the noise is emphasized. When, on the other hand, the PRF is high and the pulses closely spaced, static tends to take on the characteristics of fluctuation noise.[a] This effect is accentuated when such static is observed through a narrow-band receiver. The individual pulses are broadened by passage through the narrow-band circuit and are forced to "pile up on each other's heels," thus giving rise to an irregularly fluctuating wave instead of an irregular sequence of spikes.

The correspondence between this treatment of static characteristics and the "type" nomenclature which has been widely used in previous discussions is fairly direct. Click static is weak with a very low PRF. Lightning and local thunderstorm static is intense with low or medium PRF. Hiss static is weak with a very high PRF and perhaps should be regarded as fluctuation rather than impulse noise. In order to describe the static frequently referred to as grinders, it is necessary to describe the "burstiness" of the static in terms, for example, of the percentage time occupied by bursts and their average frequency of recurrence.

The diversity and inconstancy of natural atmospheric static make accurate measurement and rigorous specification imperative in tests which involve simulated static interference. For laboratory purposes methods of simulating atmospheric static were developed. If an audio simulation is desired, a satisfactory method[1] consists of amplifying only the peaks of the ionization noise obtained from a gas-filled tube. The center-clipping or "peak-pass" circuit can be adjusted to control the density of the noise

[a] Fluctuation noise is used as a generic term for noise characterized by highly irregular, aperiodic wave form. Shot noise, ionization noise, and thermal noise are examples.

peaks which are passed. To obtain radio-frequency [r-f] static, these audio static spikes can be boosted to high peak voltages by the action of a spark coil.[12a] The circuit is shown schematically in Figure 1. When the voltage across the secondary of the coil reaches a critical value, the spark gap breaks down and there is a very brief surge of current through the 75-ohm resistor and through any external circuits in parallel with the resistor. The average PRF can be controlled by adjusting the peak-pass circuit, and the PRF is indicated by a pulse-counting circuit at the output of a broad-band receiver.

The general utility of such a static generator recommended the development of a more versa-

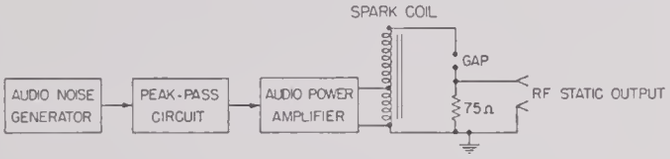


FIGURE 1. Schematic diagram of r-f static generator.

tile generator for use in calibrating noise meters and receivers, for acceptance tests of communication equipment, etc. Circuits which produce pulses of very short duration were designed and experimentally tested. A generator of radio-frequency noise was built[8] which provided steady or random PRF, the average PRF being adjustable over the range from a few cycles per second to approximately 4,000 c. The amplitude of the pulses can be either constant or random with time. Three major types of pulse generators (hydrogen thyratron, argon thyratron, and hard tube types) were investigated, and the specific difficulties with each type propose problems for further development.

13.2 RADIO VOICE COMMUNICATION

The wave form of a static pulse suggests the possibility of limiting the peak amplitude of the pulse with a peak-clipping circuit. The ability of speech to withstand severe distortion of this type (see Section 7.1) would indicate that noise-peak limiters might operate without seriously affecting the intelligibility of the speech signal.

13.2.1 Noise-Peak Limiters

Extensive articulation tests were conducted to test the effects of peak-clipping in radio receivers.[3] The tests were made with the Wasmansdorff limiter in an ARB naval aircraft receiver. The limiter was inserted in the audio section of the receiver and was adjusted so that, in the absence of interfering noise, only

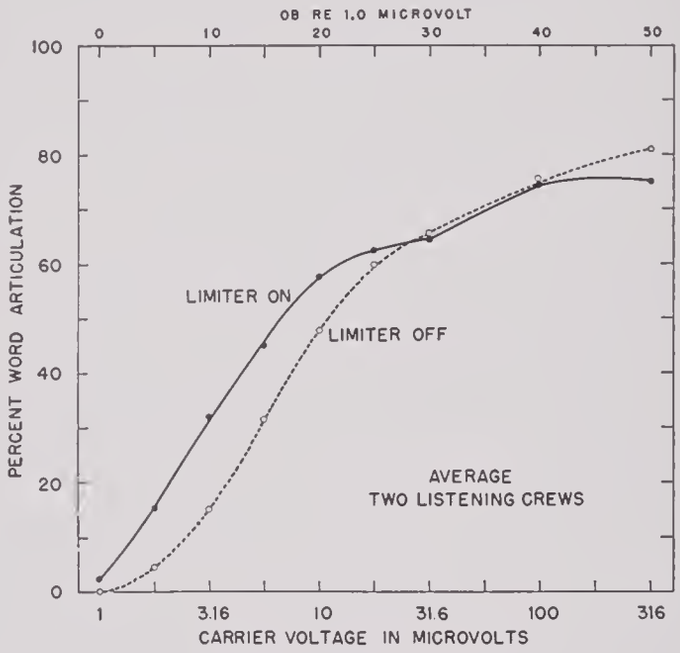


FIGURE 2. Effect of noise-peak limiter on articulation scores with static interference.

the extreme peaks of the speech signal were affected. High-amplitude pulses were clipped off at this level, and the disturbing effect of the pulse was reduced. The limiter was arranged so that it could be turned on or off in successive tests.

In tests conducted with moderately severe thunderstorm static as the only interfering noise, the limiter provided marked improvement in intelligibility, as shown in Figure 2. The increase in articulation score was as much as 15 percentage points. The improvement was restricted, of course, to the range of received signal levels in which the speech was too weak to override the interference.

Tests in quiet and tests with the listeners in

airplane noise (no static) indicated that the amplitude distortion introduced by the limiter had little effect upon intelligibility when no interference was present in the received signal. Results obtained with both the announcer and the listeners in intense airplane noise showed that the limiter was less effective when noise picked up by the announcer's microphone was present in the received signal. Even with the ambient noise interference, however, the limiter provided increased intelligibility with weak carriers in static. The maximum protection against static is obtained with noise-peak limiters when the microphone noise pickup is low.

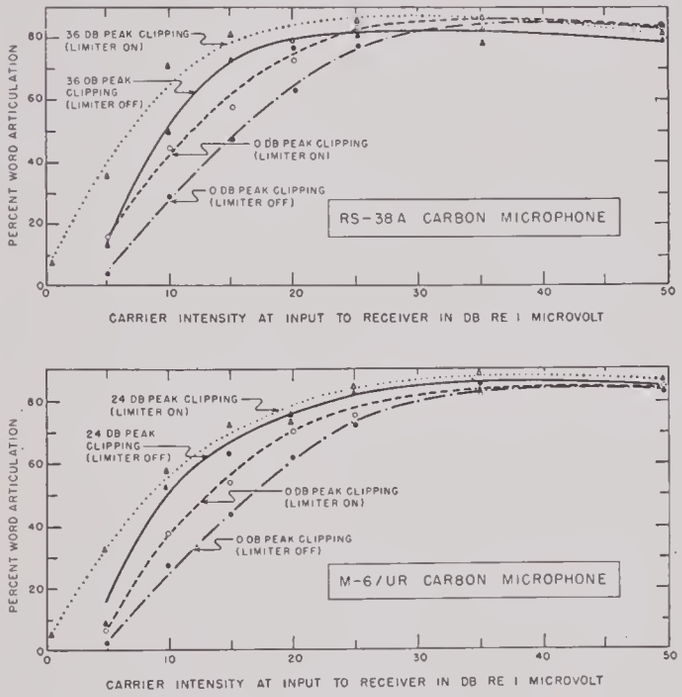


FIGURE 3. Articulation results showing the combined effects of premodulation clipping and limiting the static peaks in the receiver.

Although peak clipping has been most generally used in the radio receiver, it is also possible to clip the peaks of the speech signal prior to transmission. The advantage of premodulation clipping derives from the improved efficiency of modulation. The high-amplitude peaks of speech are removed at no loss in intelligibility, while the weak but important consonant sounds reach 100 per cent modulation.

Tests using premodulation clipping[4] showed that the intelligibility of speech transmitted by amplitude modulation [AM] of a radio-frequency carrier could be considerably improved by a voltage-limiting device adjusted to permit no more than 100 per cent modulation of the carrier wave. The adjustment of the audio gain ahead of the limiter should produce 12 to 24 db peak clipping of the speech signals, and a simple low-pass filter should be included after the voltage limiter to eliminate those audio frequencies above 5,000 c which result from peak clipping.[7] The transmission of amplitude-distorted speech, while not desirable for commercial broadcasting, is valuable for military communication where intelligibility is at a premium. Field tests supported the conclusions drawn from laboratory tests.[6]

Further tests explored the combined effects of clipping the peaks of the speech waves in an ATB transmitter and also limiting the static peaks with a Wasmansdorff noise limiter in an ARB receiver.[5] Typical results are shown in Figure 3 for two microphones and for two levels of premodulation clipping. The highest scores were obtained by clipping in both the transmitter and the receiver. Under the test conditions, the improvement over normal transmission amounted to 8 to 10 db.

Since the noise-peak limiter is incorporated in the audio stages of the receiver, the noise pulses which it discriminates against are shaped in the r-f sections of the receiver. The response of a stagger-tuned band-pass network to a delta function (pulse of zero duration, infinite amplitude, and unit area) may be approximated by two empirical relations.[10] The peak amplitude of the audio output pulse (or of the envelope of the r-f pulse) is

$$\hat{e} = B_n + \frac{0.6}{\phi} = B_n + \frac{7.5}{\Phi}, \qquad (1)$$

where B_n is the overall bandwidth measured at the half-power points, ϕ is the slope of the selectivity curve plotted in terms of relative response in amplitude units versus frequency in cycles per second, and Φ is the slope of the selectivity curve plotted in terms of relative response in decibels versus frequency. Thus, if the slope of the selectivity characteristic is held constant, the amplitude of the output pulse increases linearly with bandwidth. Or, if band-

width is held constant, the amplitude of the output pulse decreases as a hyperbolic function of the slope of the "skirts" of the selectivity curve.

The duration of the output pulses as measured at an amplitude 40 db below the peak is

$$t = 1.8\phi = 0.15\Phi. \qquad (2)$$

Thus the duration at a particular relative amplitude level is independent of the receiver bandwidth and directly proportional to the slope of the skirts of the selectivity characteristic.

The articulation results obtained in the presence of impulse noise reflect the operation of these two factors, B_n and ϕ. As successively narrower pass bands with successively steeper skirts are employed, the amplitude and power of the output pulses decrease. At the same time, the duration of the output pulses increases. If no limiter is used, the reduction of amplitude due to the narrow pass band increases slightly the intelligibility of the speech, although the effect is not as important to intelligibility as might be supposed. When a noise-peak limiter is used, the controlling factor is the duration of the output pulses. If each output pulse is limited to the same amplitude, then each pulse essentially blanks out the speech signal during a short time interval. As the duration of the pulse is made longer, the disruptive effect on intelligibility is increased. When the output pulses are long enough to overlap, the noise assumes the characteristics of fluctuation noise, and the limiter becomes ineffective. Consequently, best results are obtained with noise-peak limiters if the output pulses are of short duration, i.e., if the selectivity characteristic has gradually sloping skirts.

The effects of an amplitude selective device such as a noise-peak limiter can be summarized as follows:

1. Amplitude limiters are ineffective against fluctuation noise because there is not, statistically speaking, sufficient difference between the amplitude pattern of speech and the amplitude pattern of fluctuation noise to permit the device to favor one against the other.

2. Against impulse noise, the amplitude limiter provides marked improvement in performance if it is incorporated in a receiver with appropriate selectivity characteristics.

3. For optimal reception in the presence of impulse noise, the selectivity curve of the circuits preceding the limiter must have gradually sloping skirts.

13.2.2 Static-Canceling Circuits

Three noise-reducing circuits were developed and tested which operate on the principle that the impulse noise in adjacent radio frequencies can be used, by proper phase adjustment, to cancel out the impulse noise in the pass band of the receiver. The developmental work for two of these circuits,[2] which were intended for use with aircraft radio receiver AN/ARR-15, was done under Navy contract at Maguire Industries, Inc., and performance tests were conducted at the Psycho-Acoustic Laboratory. The third device was developed as a joint project by the Psycho-Acoustic Laboratory and Central Communications Research, Cruft Laboratory, Harvard University.

Countermodulation

The countermodulation circuit involves the use of an audio-noise voltage, provided by an auxiliary noise channel, to modulate the main signal in a special modulator incorporated into the intermediate frequency [i-f] section of the receiver. The main signal is, of course, the i-f carrier which has already been modulated by speech, and, in a sense, also by noise. The noise wave from the auxiliary channel is applied to the special modulator in proper amplitude and phase to counteract the noise component of the main signal. The result is a relatively noise-free i-f wave which can then be detected in the conventional manner.

An assumption implicit in the countermodulation scheme is that it is possible to obtain from the auxiliary noise channel a wave which contains more noise and less speech than does the signal in the main channel. This is accomplished by detuning the noise channel slightly from the intermediate frequency of the receiver, and by providing automatically adjusted bias to the noise detector or rectifier in order

to discriminate further against any low-amplitude signal component in the noise channel.

A further assumption is that the wave form of the countermodulation voltage will be approximately the same as the noise envelope in the main i-f channel of the receiver. For certain types of noise, such as interfering carriers or fluctuation noise with truly random

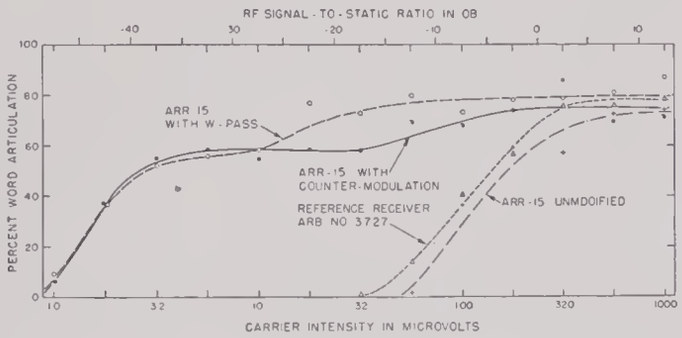


FIGURE 4. Results of tests with simulated static at an intensity of 240 μv and an average PRF of 1000 pps. The countermodulation and W-pass circuits were developmental models, adjusted on the basis of preliminary tests to give optimal performance under the test conditions. The squelch circuit of the unmodified ARR-15 (No. 6) was disabled for these tests.

phase characteristics, this second assumption is not valid, but for the impulsive variety of interference it is well justified. Modulation envelopes due to impulsive interference tend to be highly similar in adjacent frequency bands, and, if the amplitude of the interference is high relative to that of the carrier, it is entirely possible to use noise from an auxiliary channel to cancel or to countermodulate noise in the main channel.

CANCELLATION (W PASS)

The same two assumptions are basic to the cancellation or W-pass circuit. This device uses cancellation in the audio section of the receiver to remove the noise. Three final i-f amplifiers and three detectors are used. Two of the amplifier-detector channels are detuned slightly, one to either side of the main intermediate frequency, in order to pass more noise than signal. The audio outputs of the three channels are mixed, with the noise detectors 180 degrees out of phase with the main detector. When the arrangement is properly adjusted, the noise voltages from the two noise channels cancel the noise component from the main channel, and relatively noise-free speech results.[b]

Neither the W pass nor the countermodulation system should be expected to be effective against nonimpulsive interference. Even with impulse interference, the circuits cannot remove *all* the noise from the desired signal.

PERFORMANCE TESTS OF COUNTERMODULATION AND CANCELLATION CIRCUITS

The countermodulation and W-pass circuits were built as adapters, arranged to replace the circuits of the AN/ARR-15 receiver.[12] The

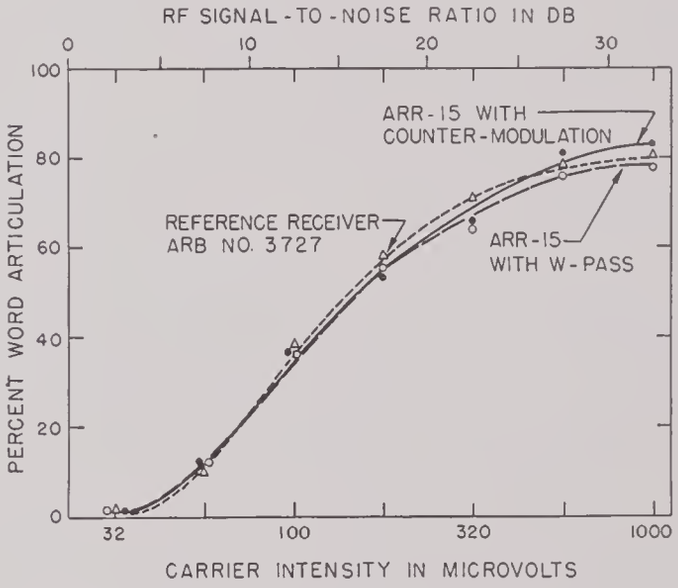


FIGURE 5. Curves showing the performance of the test and reference receivers in the presence of fluctuation noise.

method of testing involved the use of an ARB receiver as a reference or control to compare with the experimental receivers. In order to study the action of the noise-reducing circuits under a variety of conditions, articulation tests were conducted with several types of interference: irregular static pulses, regular pulses similar to the interference from a radar modulator unit, fluctuation noise, and spurious radio signals in the form of interfering carriers.

[b] Since two noise channels are used, it is convenient to use the term W pass to designate this system. This term is proposed because the two auxiliary channels, one on either side of the main channel, give to the composite selectivity curve of the receiver a form suggestive of the letter W.

Results of the tests with impulse noise are summarized in Figure 4. The ARB reference receiver is slightly superior to the ARR-15. As indicated by the figure, over 50 per cent word articulation was obtained with the noise-canceling circuits at a carrier intensity of 3.2 μv (—37 db r-f signal-to-static ratio), whereas the unmodified receiver required almost 180 μv (—2 db r-f signal-to-static ratio). This constitutes a 35-db improvement in performance

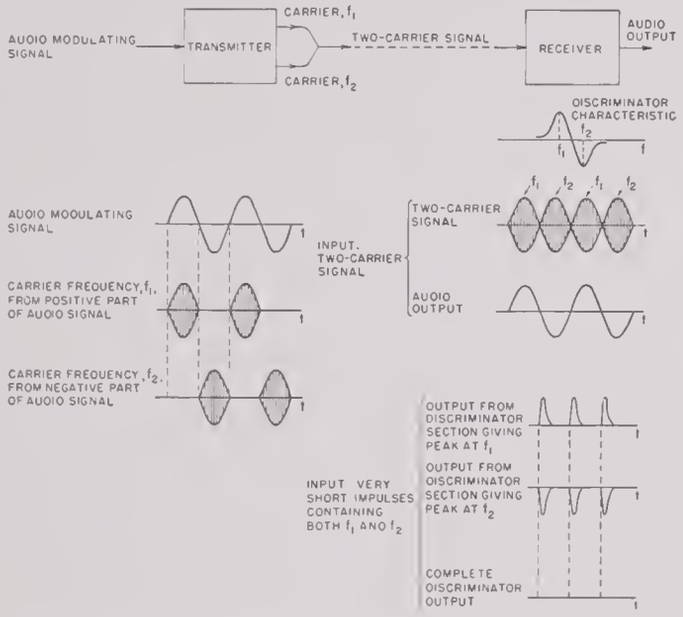


FIGURE 6. Operation of two-carrier transmission system.

due to the countermodulation or W-pass circuits.

No attempt was made to quantify the improvement provided against regular-pulse interference because there was difficulty in providing enough interference to tax the capabilities of the static-canceling circuits. It was apparent, however, that marked improvement in performance could be obtained.

The results of tests in the presence of fluctuation noise showed that there are essentially no differences in performance between the modified and unmodified receivers. These results are indicated graphically in Figure 5.

Tests with unmodulated carriers and with frequency-modulated carriers used as interference likewise showed no significant difference between the ARR-15 with noise-reducing circuits and the reference receiver.

The static-canceling circuits are, therefore, very effective against impulsive types of interference but have no effect against the other types. Since many of the serious sources of interference with military radio communications are of the impulsive type, the countermodulation and W-pass circuits hold much promise for future development and application.

TWO-CARRIER SYSTEM

This radio-communication system makes use of two carrier frequencies with a special type of amplitude modulation.[9] The two carriers are separated in frequency by a few kilocycles. One carrier is modulated in amplitude by the positive portion only of the audio modulating signal, the second by its negative portion only. In the absence of an audio modulating signal neither carrier is present. At the receiver a sort of frequency discriminator is used as a detector, with the positive and negative peaks of the discriminator response curve coinciding with the carrier frequencies. The action of this circuit is such that the original audio modulating signal is obtained from the two modulated carriers. Electric interference of a sharply peaked character affects the discriminator so that the noise is balanced out in the output. A simple limiting circuit follows the discriminator and removes any noise peaks which may remain because of imperfect balance. The result is a great improvement in effective signal-to-static ratio.

A diagram illustrating the operation of the two-carrier system is shown in Figure 6.

One of the major advantages of this system of noise reduction stems from the fact that, in order to transmit sinusoidal audio signals, just one-third the amount of r-f power is required as would be needed by the conventional AM system employing a single carrier. The system behaves in this respect much like the suppressed carrier transmission system, but without the necessity of having to reinsert the carrier at the receiver.

The major disadvantage of a two-carrier system is the broad channel required for transmission. The carriers must be separated sufficiently to reduce side-frequency interference and to make the beat frequency between the two carriers inaudible. On the other hand,

the smaller the separation of the two carriers the better is the noise cancellation provided by the discriminator. Probably 15 kc is near the optimum separation.

used to pick up a single carrier whose frequency corresponds to that of one of the two carriers for which the receiver is adjusted, and which has normal amplitude modulation. The inter-

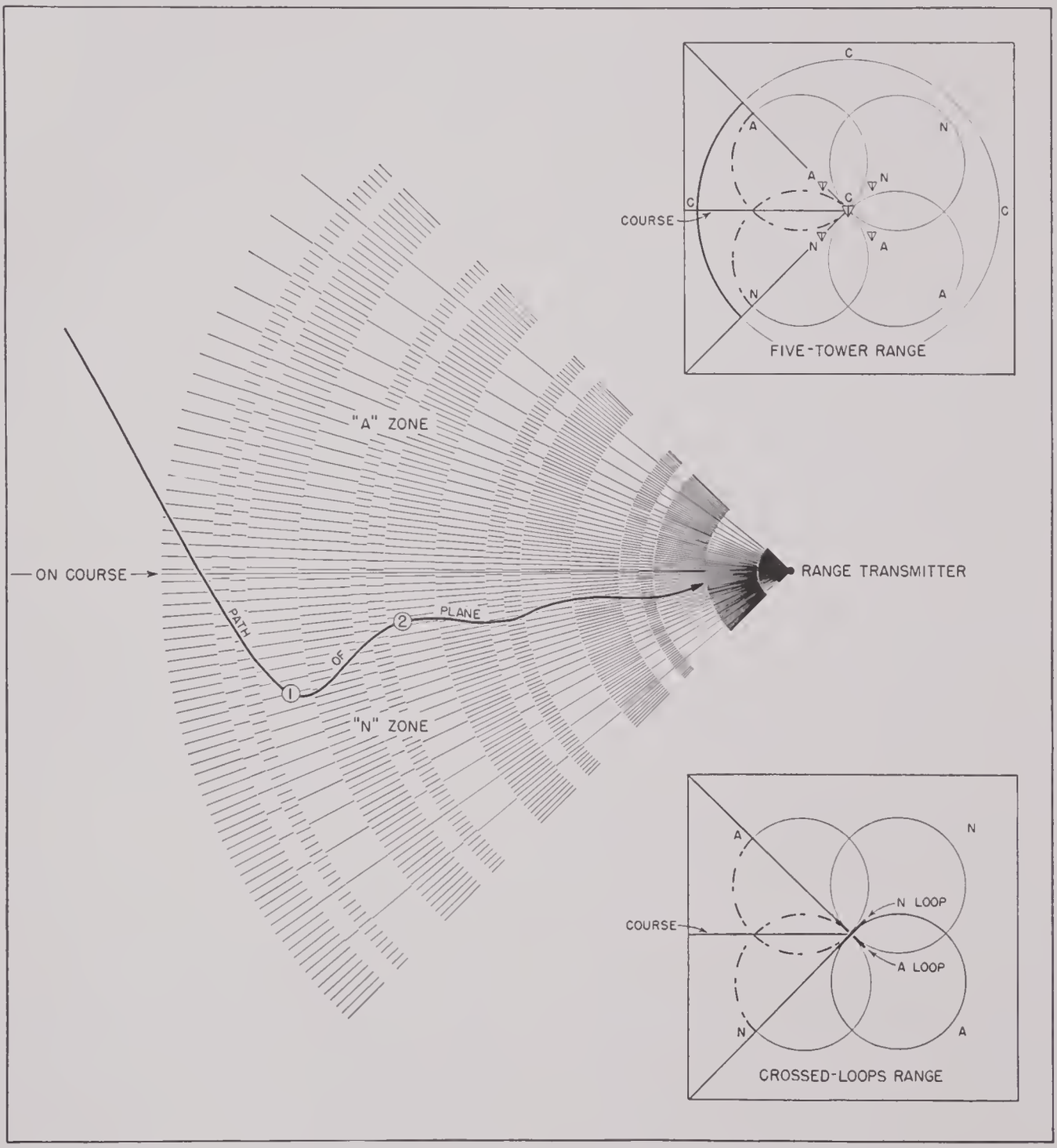


FIGURE 7. Schematic representation of one sector of a right-angled radio range.

Consideration of the two circuits will show that the two-carrier receiver is very similar in principle to the countermodulation circuit discussed above. The two-carrier receiver may be

ference reduction obtained with this single carrier is almost exactly the same as that obtained when the receiver picks up the two carriers with their special type of amplitude

modulation. Thus, the noise-reducing capacity of the system is inherent in the receiver alone. In either case, the receiver contributes about 35 db improvement over normal operation.

13.3 RADIO-RANGE SIGNALS

Since 1929, the principal radio aid to aircraft navigation has been the low-frequency radio range. In 1945 there were approximately 200 range stations, operating on frequencies between 200 and 400 kc, and providing a network of radio airway markers covering the United States.

The essential features of navigating by radio range are illustrated in the schematic representation of Figure 7. The range transmitter is located at the right-hand side of the figure, and radiates dot-dash signals (code letter *A*) with maximum intensity along the upper boundary of the sector, and dash-dot signals (code letter *N*) with maximum intensity along the lower boundary of the sector. Field strength is represented by density of shading. For purposes of illustration, a number of *A*'s and *N*'s are shown simultaneously in the range area.

The on-course line represents the line at which the *A* and the *N* signals have the same field strength, and the two signals interlock to form a steady tone of 1,020 c in the operator's headphones. Near the left-hand side of the figure the signals are weak and, therefore, difficult to make out against background noise. Near the on-course line the differential between the *A* and the *N* is low, and it is difficult to discern which signal predominates. An airplane approaching the transmitter along the line marked "path of plane" would, then, fly into the *N* zone until it reached a point (1) at which the field strength was high enough for adequate reception and the differential between *N* and *A* was sufficiently marked for the pilot or the navigator to make out the dash-dot of the *N*. Discovering that the *N* was growing progressively stronger, the pilot would correct his course and fly toward point (2). At point (2) he would again correct his course, etc. In this way he would follow or bracket the path along which he could just distinguish the *N*. This is sometimes called the "edge" of the beam, and is not, of course, sharply defined. "Flying the edge" tends to prevent collisions with aircraft flying in the opposite direction along the same course.

Of the two systems used to transmit range signals, one is the so-called "simultaneous" range which provides both range and voice signals on a single channel; the other is a simpler system which provides range signals only. The simultaneous range, also called five-tower range, is the more widely used. The simpler system employs two directional antennas, usually crossed loops.

The simulation, for laboratory tests, of radio-range signals must consider several general factors which govern the performance of radio-range systems, one of the most important of which is the human factor. The ability of the ear to distinguish small variations in the intensity of a tone, and the effects of background noise upon the accuracy of discrimination were discussed in Section 3.5. Perhaps the most important physical variables are the signal-to-static ratio, the *A*-to-*N* ratio, the density of the static, and the characteristics of the range receiver. Some of these factors, such as the differential sensitivity of the ear or the characteristics of the audio sections of the receiver, can be tested with simple audio equipment.[11] For a complete picture, however, it was necessary to use r-f test equipment. For most tests R-23/ARC-5 and R-23A/ARC-5 receivers were used.

13.3.1 Radio-Frequency Signal-to-Static Ratio and *A*-to-*N* Ratio

As a first step in studying the performance of a radio-range system, it is important to determine the ways in which the distinguishability of the received *A*-to-*N* signals depends upon the r-f signal-to-static ratio and upon the *A*-to-*N* ratio. Tests were conducted with a standard range receiver and with simulated five-tower range signals. The r-f static had an average PRF of 1,000 and an intensity of 22 μv.[13]

Results of these tests are shown in Figure

8, where the ratio necessary to distinguish A from N is plotted in decibels as a function of the r-f signal-to-static ratio. The curves, which are very similar to those obtained with audio equipment (see Section 3.5), illustrate the necessity of having a large A-to-N differential when the static conditions are unfavorable. Thus the range is essentially many degrees

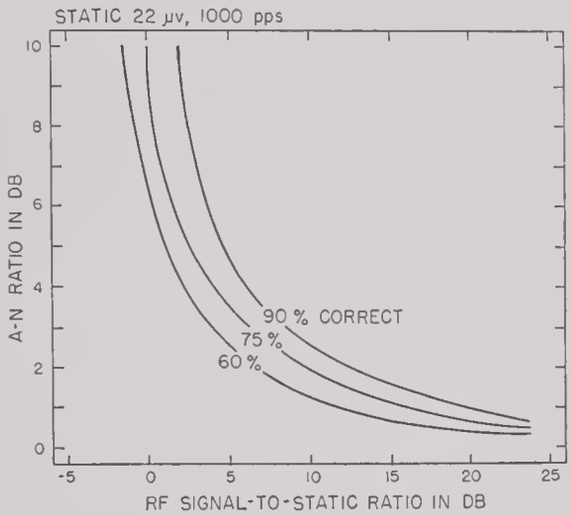


FIGURE 8. Curves showing the paired values of A-N ratio and r-f signal-to-static ratio required for 60-, 75-, and 90-per cent correct reception of five-tower range signals.

wider at a distance from the range transmitter than it is near the transmitter where the signal-to-static ratio is more favorable.

13.3.2

Static Density

The curves of Figure 9 show how markedly the actual interference value of the static is dependent upon the static density. High PRF static interferes seriously with reception, and a relatively high signal-to-static ratio is required for any particular level of performance.

As indicated by the relative positions of the curves, the interference value of the static at first rises rapidly with increasing PRF but, with further increase, a "saturation" point is approached. This saturation point is a function of the bandwidth of the range receiver and corresponds to the point at which the audio static begins to lose its spike-like wave form and to take on the appearance of fluctuation noise. For conventional range receivers, the bandwidth is 2 to 3 kc, and the critical PRF is approximately 3,000 pps.

13.3.3

Beam Filters

The simplest and most widely used noise-reducing circuit for radio-range receivers is the "beam filter." The function of the filter is simply to restrict the signal delivered to the operator's headphones to a narrow range in the region of 1,000 c, thereby eliminating a large amount of the static and little or none of the desired signal.[11a]

Radio-frequency tests were conducted with a beam filter of the type (NAF 68304) used by the Navy during the war.[13a] Results indicated that the filter is more effective against fluctuation-type noise than it is against low PRF static. When the resonant circuit of the filter is energized by a static pulse, a damped oscillation or "ringing" results at the resonant frequency. The tendency of the filter to "ring" in the presence of intermittent static may nullify any

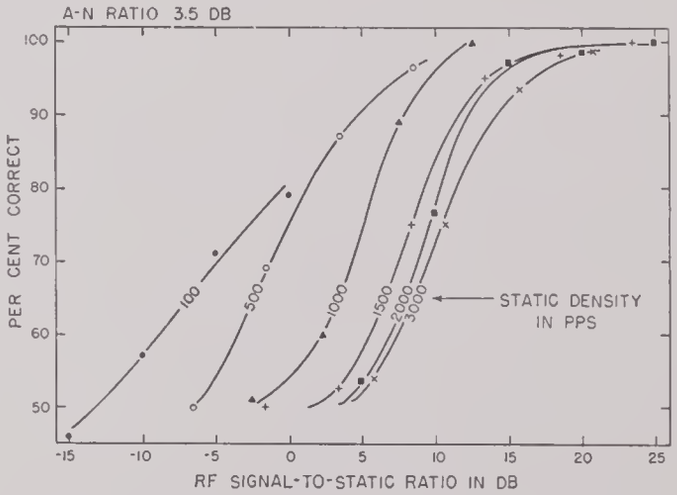


FIGURE 9. Curves showing the relation between static density (the parameter) and range performance.

possible advantage it may provide, because the ringing sound of the static is highly similar to the sound of the range signal itself. It would be dangerous, therefore, to quote a particular figure as representative of the improvement in range performance provided by a beam filter. Informal observations suggest that, in a conventional range receiver and under certain

noise conditions, the beam filter should make it possible to fly a narrow beam with about 10 db less r-f signal strength.

13.3.4 Noise Limiters

Series-diode noise-peak limiters were incorporated in two standard range receivers. Figure 10 shows the results of tests conducted with a relatively high *A*-to-*N* ratio (3.5 db).

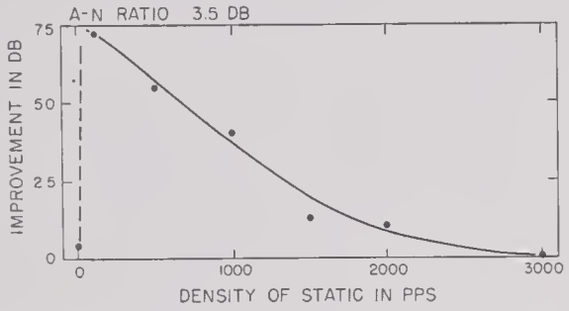


FIGURE 10. Improvement in radio-range performance provided by a noise limiter in the range receiver. The improvement was measured in terms of the reduction in the r-f signal-to-static ratio which was required for 75-per cent correct reception.

The improvement provided by the limiter is plotted as a function of the density of the static. With PRF's of 100 and 500 pps, the limiter made it possible for the listeners to record 75 per cent of the range signals correctly with 5 to 7 db less signal intensity than was required without the limiter. However, the limiter provided no improvement at all with static at 3,000 pps.

Thus the conditions under which the noise limiter showed its best performance were the opposite of those under which the beam filter had proved effective.

Inasmuch as the beam filter and the noise limiter were effective under different conditions, it appeared that they might be used with advantage in combination. It was found, however, that they did not work well together. In clipping the stronger component (for example, the *A*) of the range signal more severely than the weaker *N*, the limiter introduced stronger harmonics into the *A* than into the *N*. When no beam filter was used, the *A* had a distinctive quality which helped the listeners to distinguish it from the *N*. When the filter was introduced between the output terminals of the receiver and the headphones, however, the distinctive harmonics were attenuated. Thus one of the advantages derived from the use of the noise limiter was lost when the filter was added.

It was thought that this incompatibility might be reduced by the devising of a limiter which would clip weak signals more severely than strong ones. Such an "expander-limiter" was developed, and proved more effective than the usual type of limiter,[13b] but even its performance was not all that was expected on the basis of experience with noise limiters in communication receivers.

Tests with receivers of different bandwidths indicated that the high selectivity of range receivers (dictated by the fact that the width of range channels is narrow) is the principal factor reducing the effectiveness of limiters. This effect may be explained in terms of the pulse-transmission characteristics of band-pass circuits (see Section 13.2.1).

13.3.5 Static-Canceling Circuit

A static-canceling circuit similar to those described in Section 13.2.2 was tested in the presence of simulated thunderstorm static, and its performance was far better than that of the standard range receivers previously discussed. In order to tax the static-canceling receiver it was necessary to use static intensities above 1,000 μv, and even under these conditions the low PRF static provided less interference than the background noise of the receiver. With the static-canceling it was possible to "navigate" with signal-to-static ratios about 50 db lower than would have been required with a standard range receiver. When fluctuation noise was used, however, the performance was the same with the canceling circuit as without it.

These results for static of reasonably low PRF are suggestive of the effectiveness of the canceling principle. It is probable that, in the presence of low PRF static, equally good results might be obtained with narrow-band circuits more suitable to radio-range applica-

tions. Further evaluation must rest upon field tests or upon operational experience.

13.3.6 Other Radio-Range Devices

More or less as by-products of the study of radio-range receivers, two simple devices were developed which, when plugged into the output of a range receiver, allow the pilot to listen with greater comfort to the range signal. One of these, the radio-range signal expander,[11b] filters out the static and expands the voltage differential between the *A* and *N* signals. At reasonably large signal-to-noise ratios the range reception is considerably improved by this device.

The second circuit, the pitch modulator,[11c] converts the usual intensity variations into pitch variations, with the result that static merely raises the general pitch level or, at worst, causes a ragged pitch signal instead of noise. Thus the pilot has to distinguish the relative variation of pitch, but does not have to listen to the loud noise introduced by static in the ordinary radio receiver signal. This device is also very effective with favorable signal-to-noise ratios.

13.3.7 Reception of Voice Signals

Inasmuch as one of the important functions of the "simultaneous" range is the transmission of weather information by means of voice signals, it is important to consider the effects of the various noise-reducing circuits upon the reception of speech. One of the major difficulties lies in the high selectivity and poor audio frequency response of the receiver itself. The pass band of a conventional range receiver is too narrow for fully effective voice communication.

When voice signals are being received, a narrow band-rejection filter is customarily inserted to attenuate the range signals. Tests have shown that the actual masking caused by the 1,020-c tone is negligible (see Section 8.2), but the rejection filter may serve to reduce annoyance. However, the conventional filter eliminates a band of speech frequencies 300 to 400 c wide, and produces a decrement in intelligibility of approximately 6 per cent. The results of articulation tests conducted with audio equipment are shown in Figure 11, where the per cent word articulation is plotted as a function of the width of the rejected band for different speech-to-noise ratios. These data indicate that the filter, if used, should have the narrowest possible bandwidth.

Noise-peak limiters and static-canceling systems have been shown in preceding sections to be about equally effective for speech and for range signals. The limiter is somewhat handicapped, however, by the narrow pass band of the typical range receiver.

On the basis of these results, two avenues for future development seem feasible. First, the

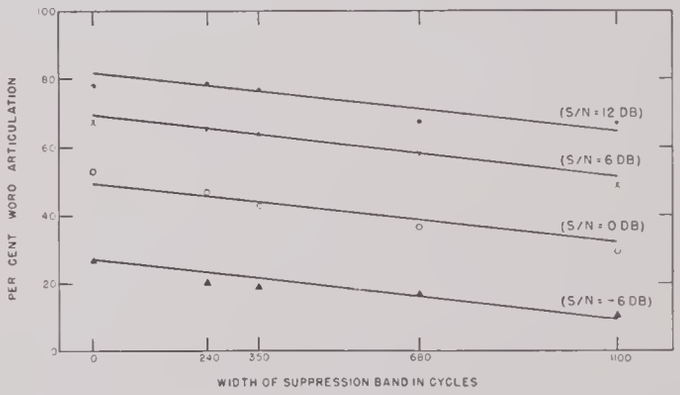


FIGURE 11. Per cent articulation as a function of the width of the suppression band.

possibility of using i-f noise limiters instead of audio limiters in range receivers should be explored. Locating the limiter ahead of some of the selective circuits should provide the advantage of wide-band limiting. Second, a narrow-band, static-canceling circuit should be developed and subjected to field tests under severe conditions of noise. Success in these two projects would ensure considerable improvement in the performance of five-tower range equipment.

Chapter 14

SELECTING AND TRAINING PERSONNEL

NO COMMUNICATION equipments are more effective than the people who use them. The finest equipment built cannot establish voice communication between a tongue-tied talker and a deafened listener. Maximum transfer of information is only achieved when the personnel as well as the equipment are the best available.

In order to obtain the most efficient personnel, a careful program of selection and training must be devised. Differences between individuals in their abilities to hear and to be heard in intense noises are surprisingly large, and the selection of the better talkers and listeners depends upon the accurate estimation of these individual differences. Also, the performance of both listeners and speakers improves after even a brief period of practice and instruction. The experience of the Psycho-Acoustic Laboratory in evaluating and training military talkers and listeners is summarized in this chapter.

14.1 SELECTION OF TALKERS

In the development of adequate articulation testing methods (see Chapter 5), it quickly became apparent that individual differences were considerable when apparently "normal" talkers spoke in the presence of intense noise. Analyses of such individual differences[1] suggested the possibility of providing relative estimates of speaking ability which would be valid for the conditions of actual military operation. As a result of these early and more or less incidental observations a separate project was undertaken to explore the problem in greater detail.

In order to obtain an adequate sample of talkers, both word and sentence articulation scores were obtained for three groups of talkers numbering 47 men in all: (1) group R, 21 talkers chosen at random from an undergraduate college population; (2) group F, 13 talkers chosen from a group of 270 enlisted men by a competent speech correctionist as exhibiting defective speech characteristics; and (3) group P, 13 talkers chosen at random from the remaining 257 enlisted men. A series of experiments were conducted, with these 47 talkers, which had as an object to determine (1) the value of subjective ratings of talkers as an index of their measured intelligibility, and (2) the basic factors underlying the individual differences and determining the intelligibility of the speech.

14.1.1 Subjective Ratings of the Intelligibility of Talkers in Noise

A number of articulation tests were conducted to provide intelligibility scores for talkers in noise, and analyses were made of the relations between the articulation scores and the scores derived from three types of subjective ratings of the same talkers: (1) preliminary ratings based on the standard speech interview for Harvard freshmen made by the instructors in an English course (Navy V-12), (2) ratings made during articulation tests by the 7 to 12 women composing the listening crew, and (3) ratings made both before and during the course of the articulation tests by two members of the Psycho-Acoustic Laboratory.[4a]

On the basis of the preliminary ratings from the standard speech interview, a group of men (group F) was singled out as exhibiting especially faulty or undesirable speech traits. As a control, an equivalent group (group P) was chosen at random from the other members of the course who had satisfactorily passed the speech interview. Both groups were tested for their intelligibility in noise, with words and complete sentences used as the speech material. For both types of material, the randomly selected group (group P) achieved a higher mean score. Although the differences between the means were significant in a statistical sense, the two groups were not nearly so different as might have been expected. On both

words and sentences, for example, some of the talkers who had been considered faulty by normal standards of judgment were among the more intelligible, and some of the randomly selected were among the less intelligible of the entire body of men tested. The overlap is shown in Figure 1 for both word and sentence materials.

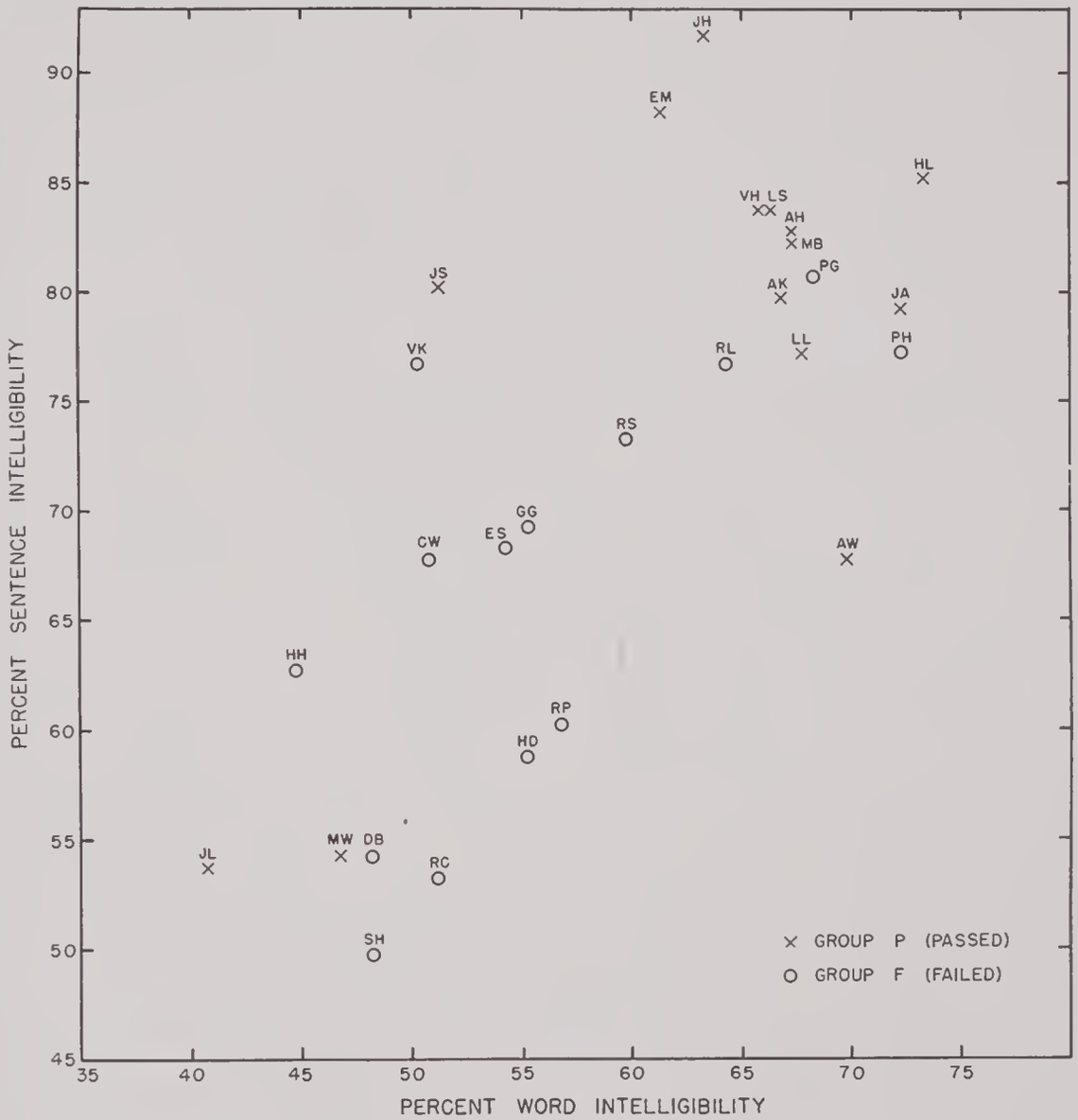


FIGURE 1. Word intelligibility vs sentence intelligibility for two groups of talkers. Group P was chosen randomly from those who had passed, and group F consisted of those who had failed a college check-list speech interview.

It was concluded, therefore, that ratings made on the basis of a standard speech interview appropriate to the requirements of a college course in speech are of limited validity for the peculiar situation of a military talker who must make himself intelligible in noise. Only a part of the criteria for superior speech in the absence of noise seems applicable in determining this type of intelligibility.

During the testing of these talkers, each member of the listening crew was asked to estimate the number of words which she had identified correctly. The listeners were never told the correct forms of the words or the sentences read, or the actual intelligibility scores achieved. It turned out that on both words and sentences the mean estimated scores showed high correlations (0.83 to 0.99) with the mean intelligibility scores. The magnitude of the correlation was maintained for various

groups of military and civilian talkers and for various types of microphones and interfering noises. After a period of practice, although the listeners had no access to information about the accuracy of their estimates, the discrepancy between estimated scores and intelligibility scores decreased until the two approximated closely.

In addition to these ratings made by the listening crew, two of the experimenters rated the intelligibility of a number of talkers on a seven-point scale. Ratings made in a preliminary interview under quiet conditions and also with noise introduced electrically into the earphones of talker and judges yielded a moderate correlation with later intelligibility scores. Other ratings by the same observers made during the articulation tests themselves showed a consistently higher correlation with the articulation score.

It would seem, therefore, that trained judges who emphasize factors found to influence the intelligibility of a voice can probably make valid ratings of a talker's capacity to make himself intelligible in noise. Should it prove in some cases inconvenient to obtain the services of experienced speech analysts, the indications are that valid estimates of the intelligibility of talkers could also be obtained from the average results of a small group of novice raters. These ratings would be based on a form of articulation test modified so as to require only a few minutes of testing for each talker. The results of the experiment indicate that the amount of success to be expected from ratings made at a naïve level without specific training in the task or knowledge of the results is fairly high. The efficiency of ratings could be improved greatly by special training.

14.1.2 Factors Related to the Intelligibility of Talkers in Noise

The second aspect of the study of individual differences among talkers was directed at the problem of understanding what it is about human voices that makes for clarity and intelligibility in the presence of noise.[4]

In order to study and analyze a group of voices whose relative effectiveness under the stress of noise was known, the 47 talkers were given both word and sentence intelligibility tests. The intelligibility scores thus obtained were used as "criterion scores" against which were correlated a variety of analytical measures made on the individual voices. Since the procedures employed require repeated measurements on each voice, there were recorded on phonograph disks several samples of both the loud and the conversational speech of each talker. From these samples measurements were made of the following physical attributes: (1) the overall intensity level of the voice, (2) the distribution (spectrum) of speech energy in the frequency range from 80 to 4,000 c, (3) the pitch of the voice (fundamental frequency), and (4) the "peak factor" of the voice (ratio of peak to effective voltage: effective voltage was read on a VU meter, peak voltage on an oscilloscope).

Also, in order to evaluate some speech attributes that are not amenable to physical measurement, five judges made systematic subjective ratings of: (1) consonant strength, (2) consonant precision, (3) appropriateness of rate of speaking, (4) duration of words, (5) steadiness of rate of speaking, (6) steadiness of level and emphasis, (7) noise penetrating quality, (8) deviation from general American dialect, and (9) overall intelligibility.

Analysis of the interrelations among the physical measurements, the subjective ratings, and the criterion intelligibility scores led to several conclusions.

When talkers are instructed to speak in "the loudest voice possible without strain" the intensity of their voices shows a low positive correlation (about 0.28) with the criterion intelligibility scores. When talkers possessing defective speech characteristics or noticeable local dialects are eliminated, however, these correlations rise to 0.45. The loudness which the talkers choose when instructed to assume a conversational level shows no correlation with the criterion intelligibility scores.

Various indices of the shapes of the individual voice spectra were devised, but no significant correlations were found between these indices and the intelligibility scores.

The pitch, or fundamental frequency of the voice, was found to vary directly with its intensity level. Any relation between pitch and intelligibility, however, was due principally to their common relation with intensity. When differences in intensity were held constant by means of partial correlation, pitch showed a slight negative correlation with intelligibility. Given two voices of equal intensity, the one with the lower pitch was apt to be the more intelligible.

Peak factors varied widely between talkers, but no correlation was observed with word articulation scores. A slight correlation was observed with sentence intelligibility.

Of the aspects of the talkers' voices which were rated, excellent agreement among the judges was obtained for ratings of overall intelligibility, noise penetrating quality, consonant precision, dialect, and consonant strength.

The average estimated intelligibility ratings on different types of speech materials correlated as high as 0.73 with the actual intelligibility scores. By multiple correlation, it was found that the overall ratings of intelligibility were dependent principally upon the variables of noise penetrating quality, consonant strength, consonant precision, and dialect.

The subjective ratings which were most closely related to the intelligibility scores were those for consonant strength and consonant precision. Absence of dialect was found to be more important for sentence intelligibility than for word intelligibility.

Contrary to expectations, estimates of the rate of talking proved of little importance in predicting the criterion intelligibility scores. However, since talker training courses often embody recommendations for the optimal number of words per minute, it is of interest to note what rate the judges considered optimal. At the conversational level, the mean number of words read per minute was 164, while the judges considered that 130 to 140 words per minute was optimal. At the loud voice level, the talkers tended to slow their reading rate considerably so that the average number of words per minute was 141. The judges still considered this too rapid a rate and their judgments indicate that 120 words per minute would be optimal when speaking in a loud voice. In Figure 2, mean ratings assigned by the judges are presented for the various rates used by the

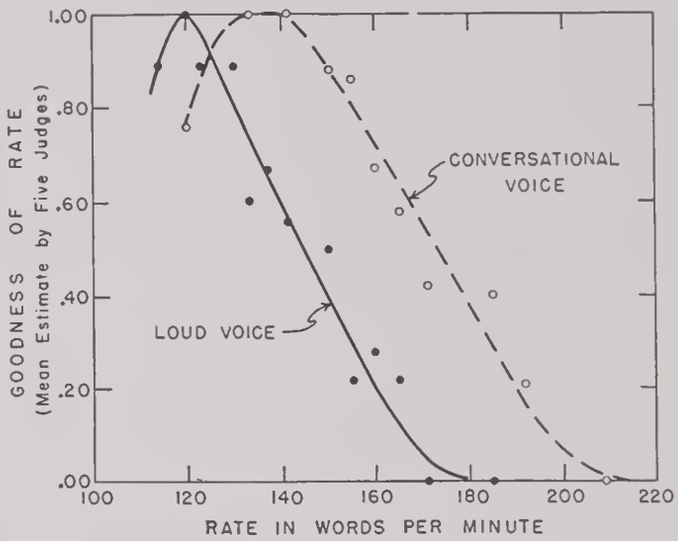


FIGURE 2. Ratings assigned by 5 judges are shown for the various speech rates used by the 47 talkers at both the loud and the conversational voice levels.

talkers. The ratings have been transformed to a common scale extending from 0 to 1. The indication is that it is possible to read too

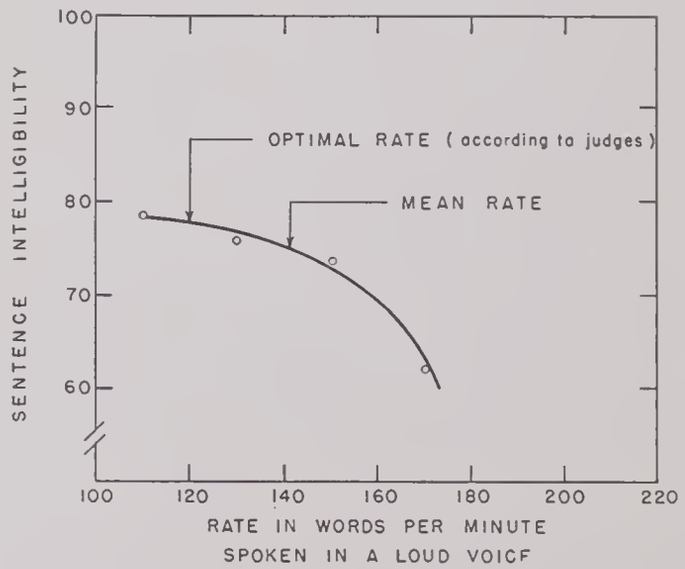


FIGURE 3. Showing the average relationship between the rate of speaking and sentence intelligibility.

slowly but that in the opinion of the judges it was not a common fault.

When the actual number of words per minute spoken by the talkers are compared with their intelligibility scores, the opinion of

the judges seems to be substantiated. There is a low negative correlation between intelligibility and speech rate, with a great deal of scatter, but if the intelligibility scores of all the talkers reading at rates between 100 and 120 words per minute, 120 and 140, etc., are averaged, the function presented in Figure 3 is obtained. Slightly better scores were made by the slowest talkers. There is little loss in intelligibility, apparently, until a rate of 160 to 180 words per minute is reached.

The practical import of these investigations for the rating and training of talkers is that intelligibility in noise can be attributed to no single voice dimension but is a complex resultant of the talkers' ability to speak loudly, to articulate speech sounds correctly and strongly, and to speak in a loud, resonant voice which seems to have the attribute of audibility through noise. The aspects of pitch, rate, duration, and steadiness were found to add little to the accuracy with which the intelligibility of the voices could be predicted on the basis of the other variables.

14.2 SELECTION OF LISTENERS

In many ways the problem of devising an adequate method of selecting the better listeners is simpler than the problem of selecting talkers. In any adequate program of talker selection some time must be spent with each individual talker, and even though a speech rating interview may be considerably abbreviated, adequate ratings of a large group of talkers is a time-consuming process. The selection of listeners, however, can go forward much more rapidly, since as many listeners can be tested simultaneously as can be seated together. Also, the criterion for listening ability may be more objectively determined than for speaking ability, since by the use of phonographically recorded speech samples, listening tests can be prepared and standardized for general use.

The ideal test of listening ability should meet several criteria. First and probably most important is the problem of validity for the military situation. The fact that individual differences in ability to hear through a variety of different types of interfering noises are highly correlated greatly simplifies this problem. As a result, it is possible to develop a test using one particular type of noise interference and to feel confident that this test will also yield a valid indication of listening ability in most other types of noise. The generality of this statement should be confirmed, however, for any unusual types of interference. It was found, for example, that the ability to hear speech through a tonal type of interference did not correlate satisfactorily with ability to hear through a random noise. The greater annoyance and unpleasantness produced by the tonal type of signal probably introduced psychological factors not sampled by the random noise test. Consequently, a separate test had to be developed for this specific application.[7, 8]

It is also important to ascertain that differences in intelligence do not affect the scores. If, for example, test words are used which lie beyond the vocabulary of the average person for whom the test is intended, items will be missed because they are not understood, even though they are clearly heard. For most tests, therefore, it is desirable to minimize the effects of intelligence on the test score. In many military situations, however, it is desirable to select listeners not only for their ability to hear speech in noise, but also on the basis of that aspect of intelligence which enables them to repeat what has been heard. Consequently, a phonographically recorded test of the listener's memory span for digits was developed.[3] While this is not a test of listening ability in the restricted sense, it may well prove useful in a battery of tests designed to select listeners for specific jobs.

As a second consideration, the listening tests must be reliable. If a test is given twice to the same group, the listeners should be rated approximately in the same way by both administrations of the test. Important in this respect is the level of difficulty of the test. In order to provide the maximum discrimination between different listeners, it is necessary to eliminate from the test those items which no one misses, and also those items which no one hears correctly. Such items are dead wood and do not discriminate one listener from another. Conse-

quently, experimentation is necessary in order to obtain a signal-to-noise ratio which provides a satisfactory level of difficulty and of test reliability.

In order to provide valid and reliable tests of listening ability, different types of speech material were recorded against a background of masking noise.[5] These tests, available on phonograph disks and suitable for group administration, were designed to be most efficient when administered over resonant earphones (TH-37 or R-14), but they have also proved to be of adequate reliability when played over a loudspeaker or over nonresonant earphones (ANB-H-1 or ANB-H-1A).

The three listening tests of value for listener selection are:

1. Words in Noise (Write-down Type)
   - Name: Auditory Test No. 2 (previously issued as Series L * Type W).
   - Items: 200 discrete words spoken to the listeners who write them down.
   - Scoring: By hand.
   - Administration time: 30 minutes for the long form (items 1 to 200); 20 minutes for the short form (either items 1 to 100 or 101 to 200).
   - Reliability: For 200 items, 0.94 (using TH-37 earphones).

2. Sentences in Noise (Instructional Diagram Type)
   - Name: Auditory Test No. 4.
   - Items: 100 commands, for each of which four drawings or diagrams are given. The listener answers by marking one of the diagrams.
   - Scoring: By hand.
   - Administration time: 25 minutes.
   - Reliability: 0.93 (using TH-37 earphones).

3. Sentences in Noise (Multiple-Choice Type)
   - Name: Auditory Test No. 8.
   - Items: 100 sentences, consisting of questions, commands, and incomplete statements, for each of which four alternative words or numerals are given. The listeners answer by underlining the appropriate response.
   - Scoring: By IBM machine or hand stencil.
   - Administration time: 25 minutes.
   - Reliability: 0.90 (using TH-37 earphones).

Each of these three tests appears to be a satisfactory measure of the ability to listen in noise. The choice between them should be based on the relevance of the speech material for the military situation, and the practical considerations of administration and scoring. Each of the tests takes about 25 minutes to administer and has been given successfully to as many as 110 men at one time (see Figure 4).

Although tests were developed using both discrete words and sentence material, there seems to be little difference between the ability

FIGURE 4. Showing two groups of subjects taking listening tests.

to interpret meaningful sentences and the ability to hear discrete words in noise. Likewise, the results obtained seemed to be independent of whether the observer is asked to write down what he hears or whether he is asked to choose the correct item from a number of different items. Further, since the score on the test depends primarily upon the signal-to-noise ratio rather than upon the overall level employed, the

scores on the tests are not significantly affected either by the variations in sensitivity among earphones of the same basic type or by the seating position of the listeners relative to a loudspeaker. For any one test, however, the listener's score depends upon the type of transducers and playback equipment used. Therefore, unless similar equipment is used at all testing centers, only the relative standings of listeners at different centers and not their absolute scores may be considered comparable.

A correlation between listening scores and achievement scores in International Morse Code after 12 weeks of training was found to be −0.17. A correlation of 0.21 was found between listening scores and intelligence for a sample of 199 listeners, where the GCT scores were used as the measure of intelligence. When the listening scores were compared with scores on the test of memory span for digits, a correlation of 0.26 was obtained, and a similar study conducted by the Psychological Corporation found correlations of 0.16 and 0.20.[2] Observations by the Psychological Corporation also gave correlations between talking and listening abilities of 0.27 and 0.41. In the light of this evidence, therefore, the conclusion seems well founded that listening in noise requires a special ability relatively unrelated to the other abilities discussed above and cannot be predicted from other such tests.

14.3 TRAINING

A carefully devised program of practice and instruction very quickly raises the level of achievement of all operators, although the most efficient performance will be attained only by training those operators who have demonstrated the greatest native ability in communication through preliminary selective tests.

The rapid and extensive improvement with practice in the performance of a group of listeners can be ascribed primarily to the operation of two factors. (1) The listeners learn to identify words partially masked by ambient noise or distorted by the characteristics of the particular equipment employed. (2) Experience in the laboratory over a long period has shown that improvement can also be expected from the simplest instructions in the use of listening equipment. The listener should be shown how to fit the earphones most directly over the ear canal and how to adjust the headband to provide the best acoustic seal against ambient noise. In addition, instructions in such matters as body tonus, fixation of attention, the major dialectical differences in pronunciation of standardized signal codes, and other devices have also been found to improve listening performance.

Speakers even more than listeners improve their performance with training. Instruction in the correct use of microphones will lead to a remarkable increase in the efficiency of communication. For example, novices in the use of hand-held microphones tend to hold the face of the microphone several inches from the lips and even experienced military fliers tell us that they habitually hold the microphone half an inch from the lips. Several experiments have demonstrated, however, that the nearer the carbon type of microphone is held to the lips without interfering with the speech movements, the greater will be the intelligibility of the words spoken. Special instructions are also needed to insure the efficient use of throat and oxygen-mask microphones. For example, it has been found that novices tend to strap a throat microphone anywhere between the jaw and the base of the neck, although even a half-inch deviation from the correct position leads to marked deterioration in intelligibility. Also, some instruction in the most effective use of the voice for the particular purpose of conveying messages in noise will improve oral communication. The variables involved here are the same as those discussed in Section 14.1.

One of the difficulties with training talkers to speak in noise is the difficulty in obtaining the proper equipment—amplifiers, noise generators, etc.—with which to simulate the usual military situation. In order to meet this need, a portable interphone for rating and training talkers in noise was developed at the Psycho-Acoustic Laboratory and subsequently placed in production by the U. S. Navy (Special Devices Division).[6] The device permits two-way communication between talkers and in-

structors either in quiet or in the presence of controlled amounts of interfering noise. The overall response of the interphone is essentially flat to 4,000 c, but filters are provided which, when switched in, introduce frequency distortion similar to that imposed by the resonant types of interphone and telephone equipment. A volume indicator makes it possible to measure levels of both voice and noise. The interphone can drive either a loudspeaker or as many as 30 pairs of standard military earphones. The complete instrument is contained in a convenient carrying case. Supplementary earphones, microphones, and cordage can be carried in a separate case. A photograph of this unit is given in Figure 5.

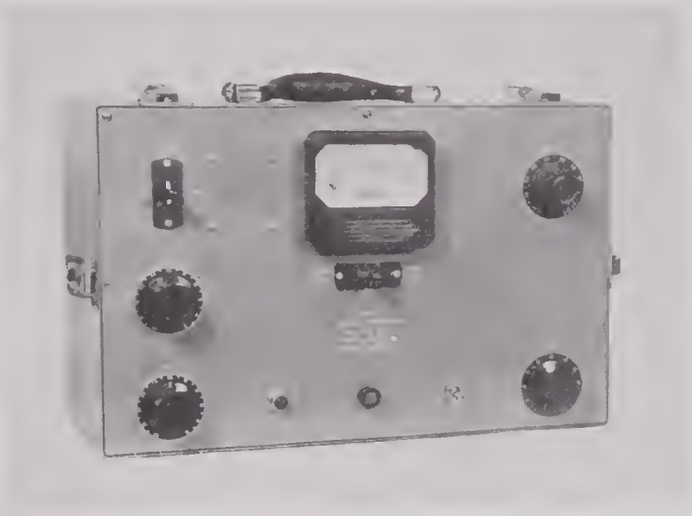


FIGURE 5. Photograph of training interphone

The unit is adaptable both for rating talkers by the interview method and for instructing and training groups of talkers in voice procedures. The interphone makes it possible to standardize and introduce precision into rating and training routines conducted by administrators stationed at different military centers. When used for classifying or rating military-telephone or radio-telephone talkers, the interphone can be used for two-way communication between the talker to be rated and any desired number of judges. The voice may then be evaluated either in the presence or absence of noise or distortion. For purposes of training military-telephone talkers, for example, an entire class can listen over a loudspeaker or over earphones, while one or two talkers demonstrate proper procedures, correct use of the voice, and the effect of noise and resonance on intelligibility. Mockups of military communication networks can be set up and students can be drilled in quiet or in noise on the use of standard voice procedures. In addition, whenever precise measurements of the intelligibility of a talker are required, the apparatus may be used to conduct standard articulation tests.

The program established at the Submarine School at New London can be taken as an example of the type of training program which is possible.[9] In this particular instance, it was recommended that the training be given at three levels: (1) a basic course to teach the general skills in the use of telephone equipment, the ability to talk intelligibly over noise, and the observance of correct voice procedures and circuit discipline; (2) an intermediate course to give each man proficiency in the use of the particular phraseology required by his assignments aboard ship; and (3) an advanced course to drill the crew as a combat team in the coordinated use of the communication circuit.

In the course of developing this program it was concluded that a short basic course (4 to 6 hours) emphasizing correct enunciation, standard phraseology, and circuit discipline, produces a marked improvement in the ability of individuals to communicate intelligibly. More important, however, in increasing the overall efficiency of communications aboard ships are the intermediate and advanced courses. It is in these phases of the training that the application of standard procedures to specific jobs and the coordination of communications for the crew as a whole are learned. Thus, the specific details of any given training course would necessarily be designed to meet the particular requirements of the services involved. In general, however, it may be said that the application of the principles discussed above within the framework of the particular situation can be expected to produce rapid and extensive improvements in communication performance.

Chapter 15

HEARING AIDS

THE PURPOSE of a hearing aid is to assist the hard-of-hearing patient to make the best use of his residual hearing. Thus a hearing aid should deliver to the patient's ear speech sounds sufficiently amplified and of appropriate quality to be intelligible in spite of his hearing loss. It should operate effectively over a wide range of input levels, without exposing the patient to possible discomfort from excessively loud sounds or to unnecessary annoyance from extraneous noise.

The discussion of hearing aids can be divided into four related problems: (1) the measurement of hearing loss, (2) the evaluation of hearing-aid performance, (3) the selection of a hearing aid for the patient, and (4) the design recommendations for future hearing aids.

15.1 METHODS OF MEASURING HEARING LOSS

Any "rating" of auditory function by a single number will always necessitate some simplifying assumption, either in the construction of a simple test or in the interpretation of an elaborate test. At least two very different requirements are involved in the objectives of different auditory tests. One is the selection of listeners where the objective is to be sure that all who pass the test have at least "normal" auditory function over a wide band of frequencies. The other is the detection and evaluation of impairment of hearing for speech to determine the necessity for special attention, for discharge from service, or for aural rehabilitation. It would be fortunate, indeed, if a single test sufficed for both purposes. Theoretically, the sensory functions involved might perfectly well differ as much from one another as visual acuity differs from color blindness.[8]

Because of its value to the diagnostician, the pure-tone audiometric test has for many years been the accepted test of hearing loss. In many cases, however, it is valuable to have a direct estimate of the hearing loss for speech. Two tests, identical in design and in method of administration, have been developed by the Psycho-Acoustic Laboratory for this purpose. The tests differ only in the nature of individual items. They are phonographically recorded, and may be administered to a single subject or to a group of subjects having similar audiograms.

An important characteristic of the two tests is that the items are homogeneous with regard to difficulty for normal ears. Homogeneity is important for two reasons. (1) The structure of the test is such that the items are divided into subgroups and each subgroup recorded at progressively lower levels of intensity. Thus, in rescrambled versions of the tests, the basic difficulty is not changed. (2) When the items are essentially homogeneous, all the items are understood within a relatively narrow range of intensities above the threshold of detectibility. This is a desirable feature, since it insures that the function relating the percentage of the items correctly heard to the intensity at which they are presented will be steep, and an arbitrary threshold for the test can be expressed in decibels with a minimum of variability.

Auditory Test No. 9[4] consists of two lists of 42 dissyllabic words of the spondaic stress pattern, i.e., words such as "railroad" and "greyhound," in which both syllables are accented. The words are divided into seven groups of six words each, and each group is recorded at progressively lower intensity levels 4 db apart. Auditory Test No. 12[5] consists of eight lists of short, simple questions. Each list is composed of 28 items which are divided into seven groups of four items. Each group of four items is recorded at an intensity level 4 db lower than the preceding one. In the selection of items for Test No. 12 an effort was made to have the sentences short, simple, representative of English sounds, typical of conversational speech, and homogeneous with respect to recognizability for normal listeners. Homogeneity for the sentences was achieved by determining the threshold of intelligibility for all the sentences

and then recording each sentence at a level so adjusted as to equate all the sentences in the group for average intelligibility.

The hearing loss for any particular subject is taken as the difference between his threshold of hearing on either of the two tests and the average threshold established under the same conditions for a group of normal subjects. As long as the same playback system and testing situation are maintained, it is possible to determine a relative intensity value for the threshold of hearing for speech. This objective may be achieved satisfactorily even though nothing is known about the "absolute" speech intensities involved.

The obvious shortcoming of such a speech test of hearing loss is that, taken by itself, no indication is given as to whether the hearing loss is due to inability to hear high frequencies or low frequencies. An experimental attempt was made, therefore, to show a diagnostic difference between uniform and high-frequency losses for speech by the use of filters in the speech circuit. The words used in Auditory Test No. 9 were passed through a high-pass circuit which cut off at 4,000 c and filtered the frequencies below 4,000 c at the rate of 17 db per octave. Since the frequencies below 4,000 c are considerably attenuated, the listener is forced to use the higher frequencies of the speech in order to interpret the sounds. This test was given to both normal and high-tone hard-of-hearing subjects. The preliminary experiments showed that persons whose audiograms are flat, regardless of the degree of loss, suffer the same loss on the filtered as on the unfiltered version of the test. However, patients who have hearing losses primarily for the high frequencies hear fewer words with the filtered than with the unfiltered form of the test. One of the chief troubles with this test in its present form is that, although the words used are homogeneous in their undistorted form, their homogeneity decreases sharply when they are put through a high-pass filter. Consequently, a much wider range of intensities is needed to cover the range from zero to maximum intelligibility, and greater variability is obtained.

15.2 EVALUATION OF HEARING-AID PERFORMANCE

A major part of the investigation of hearing loss was concerned with the experimental study of hearing aids, both as physical instruments and as aids to hearing. The physical aspects were studied at the Electro-Acoustic Laboratory and the psychological aspects at the Psycho-Acoustic Laboratory.

15.2.1 Electro-Acoustic Measurements

Ideally, the performance of a hearing aid as measured by physical laboratory tests should represent the performance of the aid when worn by an average person under normal conditions. At the same time, it is highly desirable that laboratory test methods and equipment be as simple and easily duplicable as possible. In principle, it is feasible to make meaningful physical measurements of the performance of hearing aids worn by human talkers and listeners. The concepts outlined in Chapter 9 for interphone systems could be applied and performance could be measured relative to the orthotelephonic reference system. Physical measurements of this type are, as a rule, neither simple nor easily performed, and since the conditions of use of a hearing aid vary widely, a comprehensive testing program of this type would be impractical.

The physical test methods for hearing aids now in common use are the result of a compromise. They involve, in general, the mounting of the hearing aid in a known sound field produced by a loudspeaker (artificial voice) in a room free from acoustic reflections, and the measurement of the characteristics of the sound pressure developed in a closed cavity (artificial ear) by the hearing-aid earphone.

These methods, along with the results of measurements on a large number of hearing aids, are described in an extensive report.[7] The methods used represent a combination of procedures developed at the Electro-Acoustic Laboratory with those discussed in a report[3] and those established by the Committee of the American Hearing Aid Association.[6] They have the advantage of measuring the performance

of the aid under simplified conditions which can be easily standardized. On the other hand, conditions of use are approximated only in a general way. Consequently, additional experiments must be performed to determine the corrections to be applied to the laboratory results to take into account the effect of conditions of use and to cover the range of variability which may be expected. A number of such experiments are discussed at the end of this section.

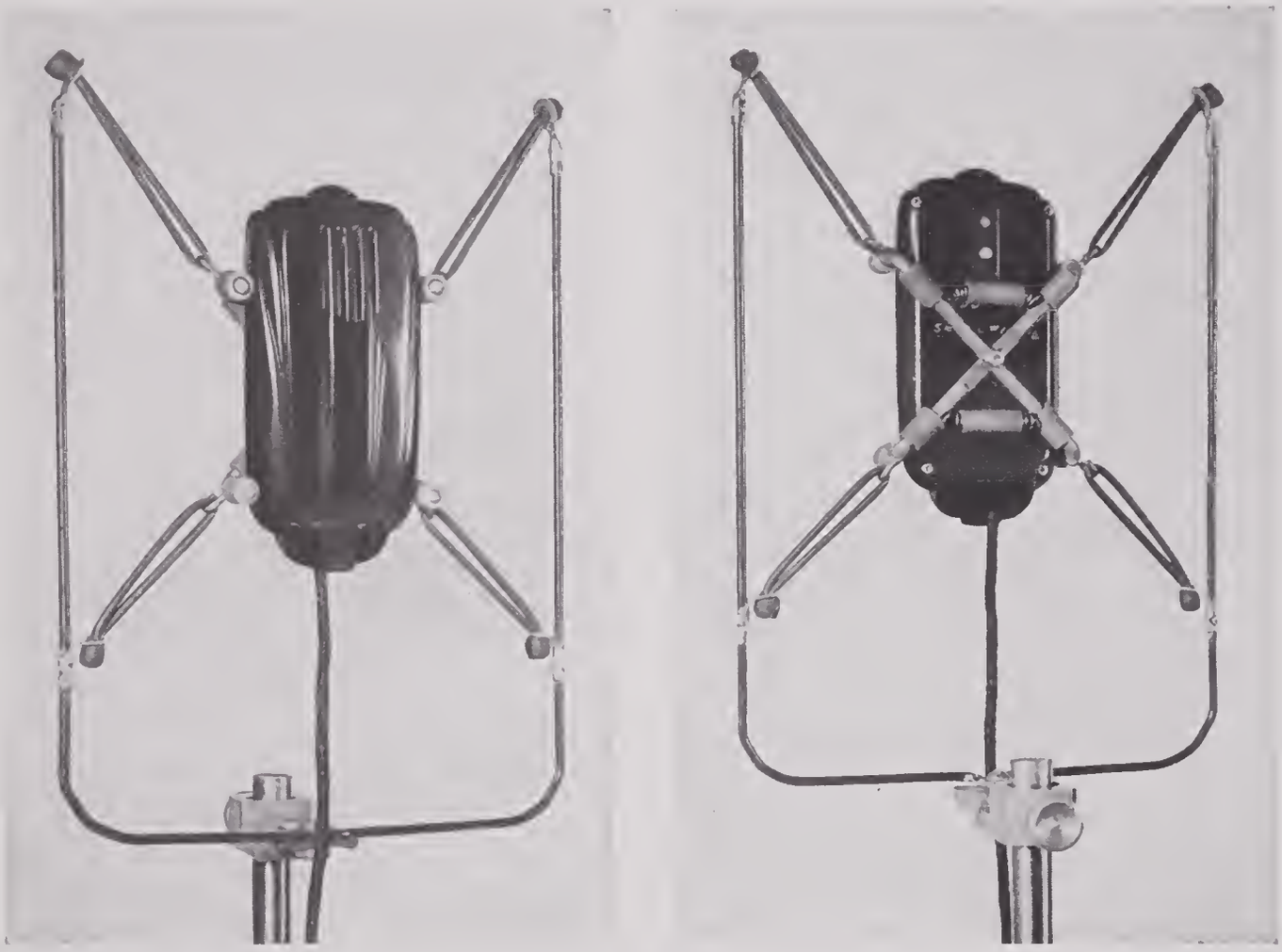


FIGURE 1. Clamp for electro-acoustic tests on hearing aids.

Present-day, wearable hearing aids of the air-conduction type[a] are generally of either the carbon or the vacuum-tube variety. The *carbon aid* is identical in principle with the ordinary telephone. It consists of a carbon microphone (transmitter) and a magnetic earphone (receiver), with operating power supplied by portable batteries. A carbon amplifier (a receiver coupled to a carbon transmitter) may be inserted between the microphone and earphone to provide increased amplification. The *vacuum-tube aid* consists of a microphone, usually of the crystal type, an electronic amplifier, and an earphone, usually of the crystal or magnetic type. Portable batteries supply the operating power.

Against the somewhat decreased weight, lower battery requirements, and greater simplicity of the carbon aid, the vacuum-tube hearing aid has the following advantages.

1. Larger gain.
2. Increased frequency range.
3. Better and more flexible control of the frequency-response characteristic (tone control).
4. Better stability.
5. Absence of noise inherent in carbon-type microphones.

It is for these reasons that the vacuum-tube

[a] Bone-conduction aids do not form part of this discussion.

air-conduction hearing aid is now in most extensive use.

In the laboratory, the hearing aid under test was mounted in a small scissors-type clamp which was elastically suspended by rubber bands from a wire bent into the shape of a U (see Figure 1). Clamp and aid were located facing a loudspeaker at a distance of 1 m in a room free from acoustic wall reflections [anechoic chamber (see Chapter 10)]. The loudspeaker (artificial voice) was either a Western Electric Type 751-B (see Figure 2) or a Western Electric Type 750-A unit (see Figure 3) mounted in a long tube lined with absorbent material. The response characteristic of one of these units is given in Figure 4. Driving voltage for the loudspeaker was obtained from a power amplifier, driven in turn by an oscillator whose frequency could be varied over the audible frequency range.

FIGURE 2. Artificial voice (WE Type 751-B).

The earphone was located in a control room adjacent to the anechoic chamber and was coupled to a rigid cavity of about 2 cu cm volume which acoustically simulated a human ear into which a conventionally molded Lucite earpiece had been inserted.[3a] The sound pressure developed by the earphone in the cavity was measured by a sound-pressure meter (see Chapter 10) using a Western Electric Type 640-A or 640-AA condenser microphone. A cross section of the coupler used by the Electro-Acoustic Laboratory is shown in Figure 5.

The same sound-pressure meter and microphone were used to set up a known free-field sound pressure, p_1, in the anechoic chamber at the location of the hearing aid. If the hearing aid is then introduced into the sound field and the sound pressure p_2 is measured in the artificial ear, the *overall acoustic gain* of the aid is defined as

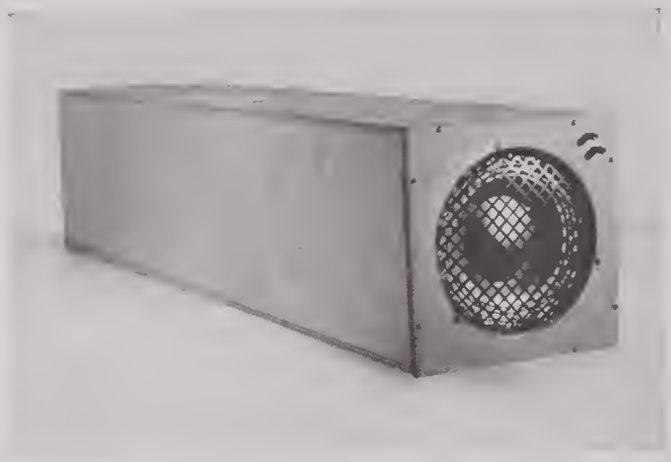


FIGURE 3. Artificial voice with acoustic termination (WE Type 750-A).

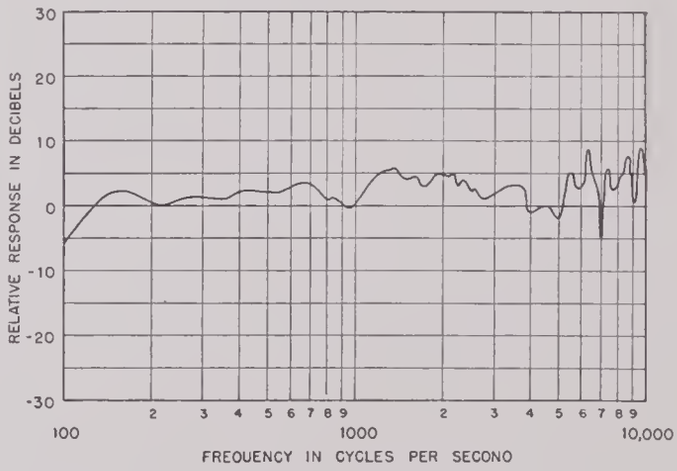


FIGURE 4. Frequency-response characteristic of artificial voice.

$$\text{Acoustic gain in db} = 20 \log \frac{p_2}{p_1}. \qquad (1)$$

This procedure is schematically illustrated in Figure 6. With this general setup, more than thirty commercial hearing aids underwent the battery of tests briefly described below.

FREQUENCY-RESPONSE CHARACTERISTIC

With a free-field input sound-pressure level

of 60 db[b] the gain control of the aid was adjusted so that a sound-pressure level [SPL] of about 100 db was reached at the frequency of maximum acoustic gain. In order to simplify comparison of the shapes of response characteristics of different hearing aids, the curves were plotted relative to the response at 1,000 c. This was done for various representative settings of the tone control.

Output versus Input Characteristic

To determine the range of linear operation of the aid the output sound-pressure level was determined for input sound-pressure levels of

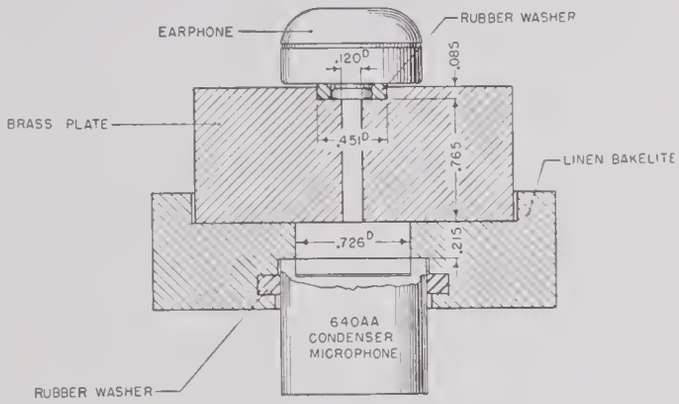


Figure 5. Artificial ear (volume 2 cu cm).

40, 50, 60, 70, 80, and 90 db and maximum gain-control setting. This was done for the following frequencies: 600, 1,000, 2,000, 3,000 c, and the frequency of maximum acoustic gain. The output sound pressures are plotted against the input sound pressures, with frequency as the parameter.

Nonlinear Distortion Characteristics

The gain control was set corresponding to an acoustic gain of 20 db and a free-field input sound-pressure level of 60 db was established. The output voltage of the artificial ear was analyzed by a narrow-band harmonic analyzer. From measurements of the fundamental and the harmonics, a single figure was computed in the conventional manner as the measure of harmonic distortion. Various volume-control settings were used with a range of input levels up to 90 db, at frequencies of 600, 800, and 1,000 c.

Miscellaneous Tests

The following tests were also performed: variation of gain with battery voltage, measurement of battery drain, and effect of temperature and humidity.

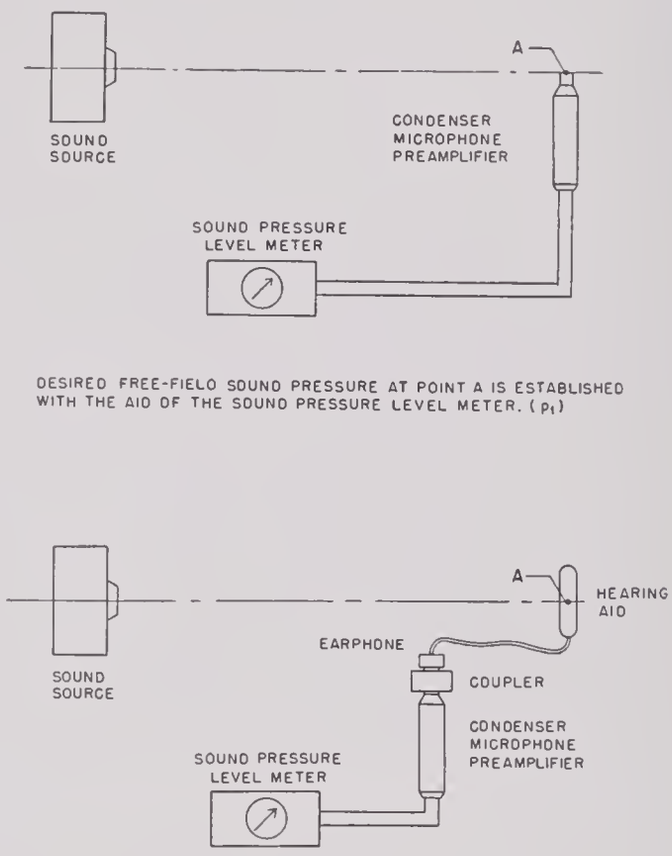


Figure 6. Measuring the overall acoustic gain of a hearing aid.

Tests of Component Parts

It is frequently desirable to know the acoustic and electric characteristics of the microphone, amplifier, and earphones as individual units. The techniques of measurement are very similar to those described above for the overall characteristics of the complete hearing aid. In the case of the microphone, the voltages developed by the unit for various known free-field sound-pressure inputs are measured. Measure-

[b] All sound-pressure levels are expressed in decibels with reference to 0.0002 dyne cm^2 which is the reference level standardized by the American Standards Association.

ments of amplifier characteristics are purely electrical and present no special problems. For earphones, the sound pressures developed in the artificial ear are determined for specified electric inputs to the earphone. Two useful ways of specifying the electric signal to an earphone are either in terms of constant voltage or in terms of 1 mw constant power available (see Chapter 10). The absolute value of the electric impedance of the earphone, terminated acoustically by the artificial ear, is also a useful quantity.

Conclusions

From the extensive physical tests of commercial hearing aids conducted at the Electro-Acoustic Laboratory, the following general conclusions can be drawn.

1. The frequency-response characteristics are peaked. The most prominent peak is usually caused by the earphone characteristic and occurs in the frequency region from 1,000 to 2,500 c. Secondary peaks may arise from both earphone and microphone characteristics.

2. Tone controls are often ineffective, acting only as an additional gain control without markedly affecting the shape of the response characteristic.

3. Nonlinear distortion generally decreases with increasing frequency of the signal for given conditions of input signal and volume control.

In Figures 7 to 9 characteristics of three representative hearing aids are given; two are of the vacuum-tube type and one of the carbon type.

Performance Characteristics of Hearing Aids under Conditions of Use

The physical tests outlined above can be used to good advantage for comparison of different makes of hearing aids. They yield only a rough approximation, however, to the characteristics obtaining under "normal" conditions of use. A series of supplementary measurements are described below which are useful in determining the nature and range of the effects caused by the difference between laboratory and use conditions. The principal differences are due to the following facts.

1. The artificial ear does not duplicate acoustically the human ear with earpiece.

2. The fit of individually molded earpieces to the wearer's ear and to the earphone varies.

3. The body of the wearer acts as an acoustic "baffle."

4. Sometimes the microphone of the aid is covered by clothes.

5. The acoustic environment and the sound fields to which the wearer is exposed are different from the sound field in the anechoic chamber.

A direct comparison of the response characteristics of various earphones on the coupler and on the ear may be made if means are provided for measuring the sound pressure which the earphones develop in the ear canal at the tip of the earpiece with which the earphone is worn. This is done by attaching a probe tube to the Type 640-AA condenser microphone of the sound-pressure meter with suitable coupling rings. The probe tube was molded into the earpiece with its opening just at the edge of the channel transmitting the sound from the earphone (see Figure 10). The probe tubes used were about 3 to 4 in. in length, with a bore of 0.025 in. Considerable attenuation occurs when sound is transmitted through the tubes, and each tube must be calibrated individually. A typical correction curve to be applied to the calibration of a condenser microphone when used with a probe is shown in Figure 11.

To obtain reproducible conditions of good seal, the earphone was sealed to the earpiece which in turn was sealed into the subject's ear with a mixture of beeswax and lanolin. The same earphone was also tested on the coupler. The ratio of the sound pressure developed in the ear to the sound pressure developed in the coupler is plotted, in decibels, in Figure 12 for three earphones. The results have been averaged for three subjects and have been labeled "coupler correction" for brevity. It is clear that this correction is a strong function of the type of earphone.

Figure 13 shows the average ratio (leakage correction) of the sound pressure as measured by the probe tube with and without the sealing agent (lanolin and wax). The results of several

earphones were averaged and plotted for three observers. If earpieces are used which are not individually fitted to the wearer, considerably greater values of leakage correction are to be expected.

The response of a hearing aid placed in a given sound field will also be affected by the presence of sound-reflecting surfaces nearby. In the standard laboratory tests, described at the beginning of this section, care was taken that there be no such disturbing surfaces. In actual use the hearing aid is worn on the per-

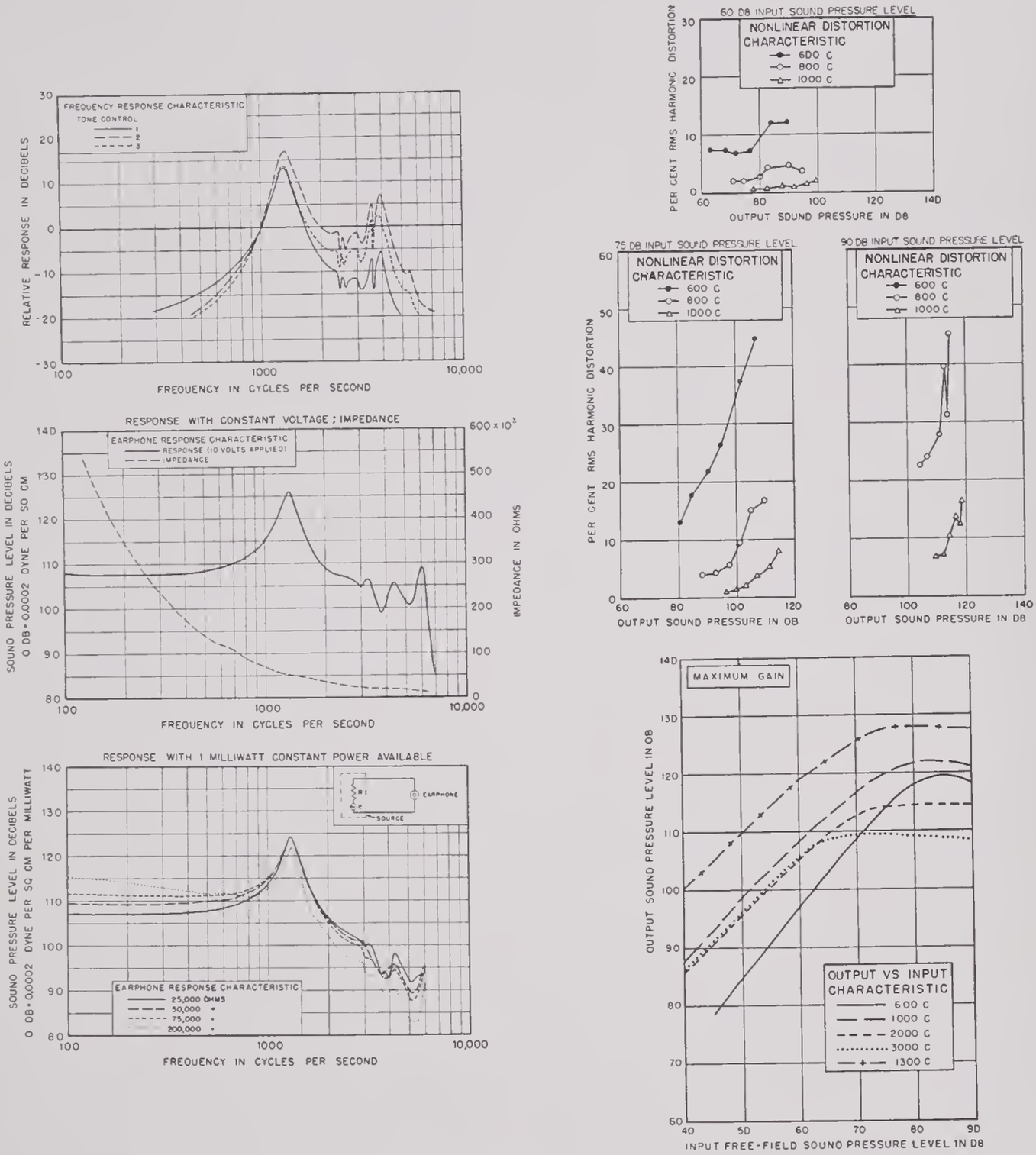


FIGURE 7. Performance characteristics of hearing aid X.

son, who presents a sizable reflecting surface or acoustic "baffle" directly in back of the aid. The position on the person in which the aid is worn, the size and shape of the individual, the nature of the clothing worn, etc., will affect the response characteristic.

Measurements were made to determine this body-baffle correction by measuring the response of a given aid in the free sound field and with a number of subjects wearing it. The acoustic effect of the human body can be broken down into three parts. First, there is an increase in response of about 5 db at low frequencies, falling off to zero between about 600 and

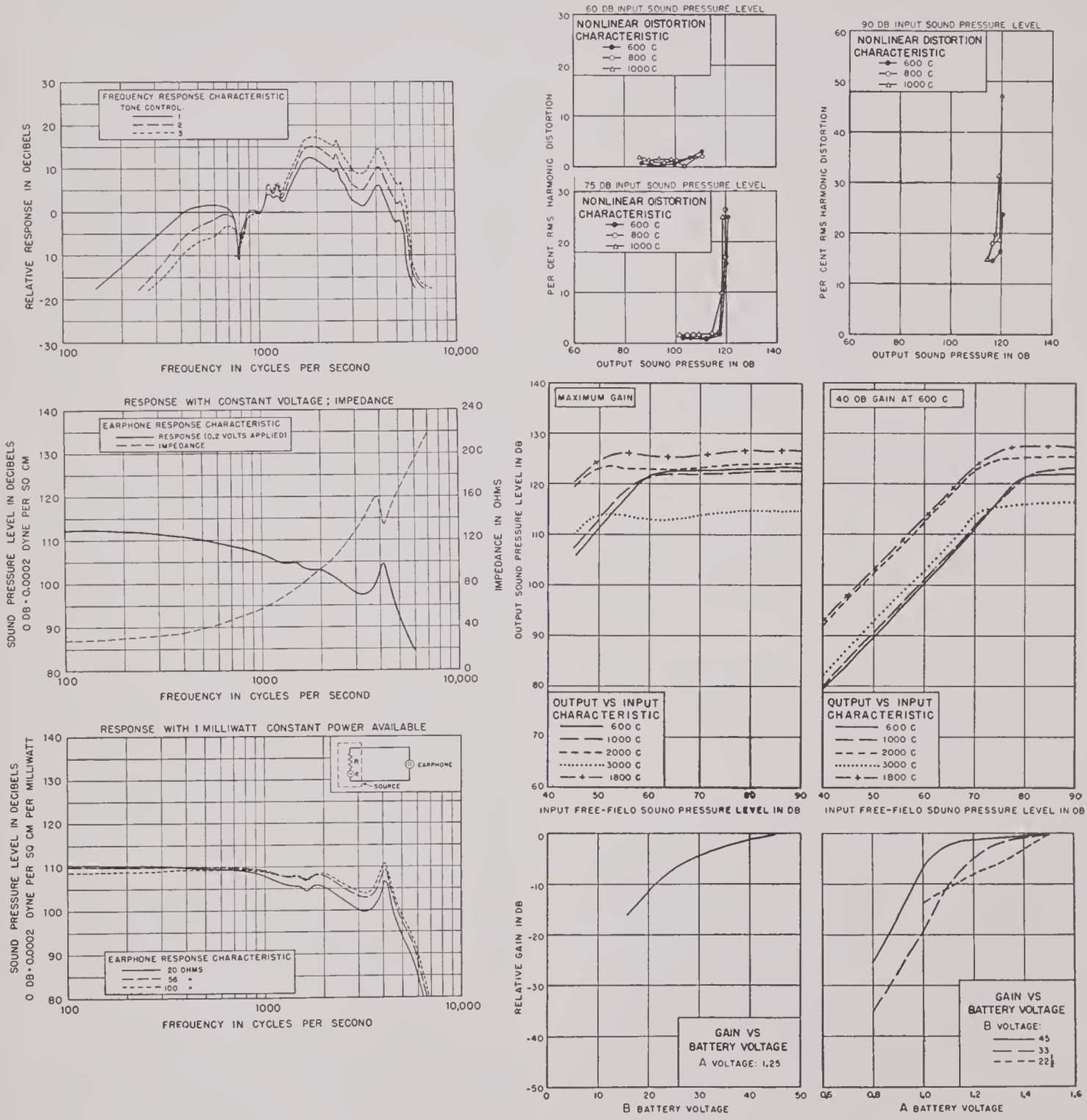


FIGURE 8. Performance characteristics of hearing aid Y.

1,000 c. Second, there are one or two marked dips in the frequency range between 1,000 and 2,500 c. This dip is a strong function of the

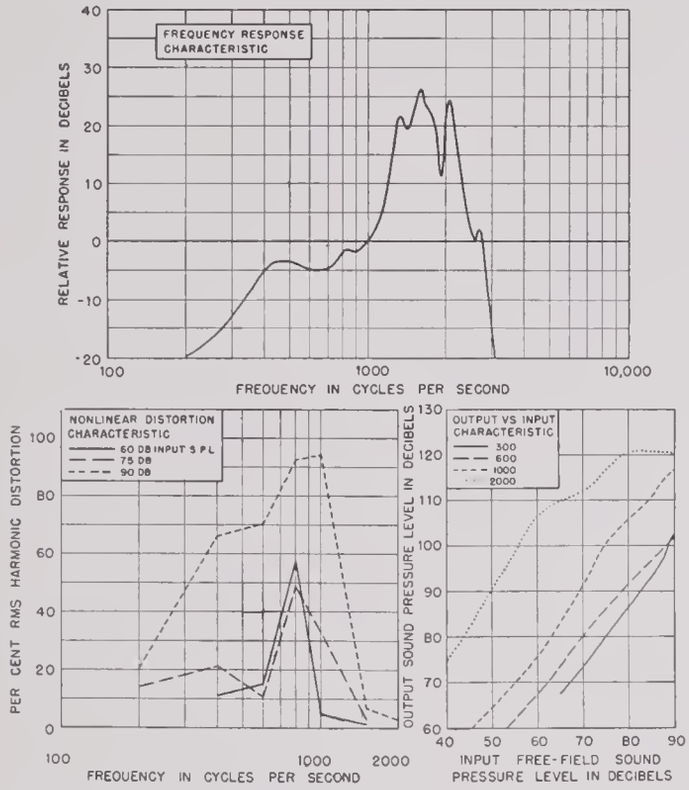


FIGURE 9. Performance characteristics of hearing aid Z.

spacing between body and aid. Third, the response above about 2,500 c is not affected significantly. Figure 14 shows typical results for a given aid and four subjects.

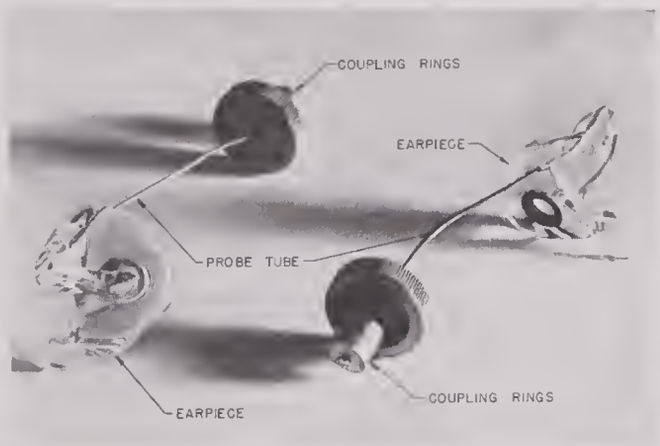


FIGURE 10. Molded earpieces with probe tubes.

Figure 15 shows typical corrections for the case where a suit coat is worn over the aid.

To investigate the magnitude and character of the acoustic body-baffle effect of the wearer's person on the response of a hearing aid in more detail, measurements of this effect were also carried out in a "random" sound field. The subjects were located in a room with hard walls

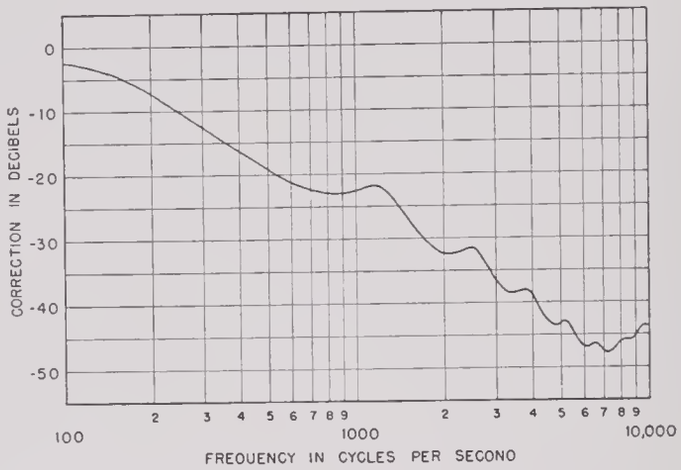


FIGURE 11. Typical microphone correction for probe tube.

equipped with polycylindrical protrusions to insure that sound waves of about equal intensities impinged on the wearer from all directions.

The body-baffle effect so determined was of considerably smaller magnitude than the one determined in the anechoic chamber. These two conditions of acoustic environment encompass

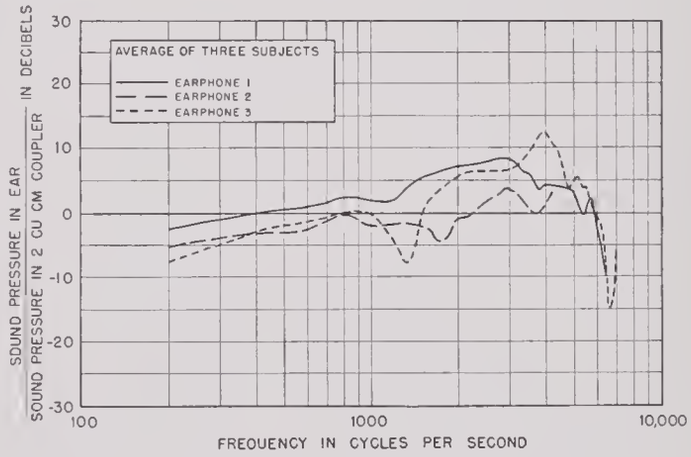


FIGURE 12. Coupler correction.

about the range encountered in use. Tests conducted at the Psycho-Acoustic Laboratory indicate that the body-baffle effect is not important, insofar as the intelligibility of speech is concerned. Hence it can be concluded that the body-baffle effect measured under free-field conditions is not representative of most conditions of use.

The effects of environment are perhaps the largest correction to be applied to the response characteristics obtained by the physical methods outlined at the beginning of this section.

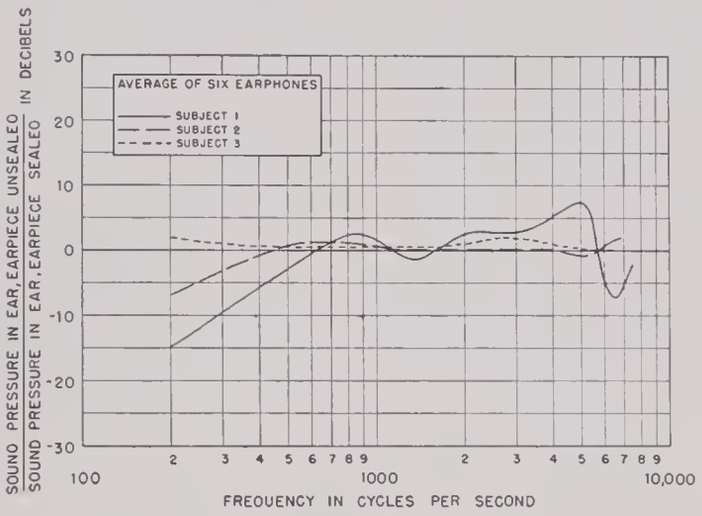


FIGURE 13. Leakage correction.

These measurements, as will be recalled, were carried out in the very specialized acoustic environment of the anechoic chamber. This was done in order to investigate the electro-acoustic performance of the instruments in a known and easily specifiable sound field. Under actual conditions of use, however, the instrument will usually be worn in rooms which are not highly soundproofed and, furthermore, the wearer of the aid may very frequently not be facing the source of sound to which he is listening.

It should be emphasized that in normal use the character of the received sound will be strongly dependent on the acoustical characteristics of the surroundings. At present no data are at hand which indicate what the differences may be between the response characteristics measured by the standard laboratory procedure and the apparent characteristics when the hearing aid is worn in an "average" room or office.

The difference in performance of the hearing aid under laboratory conditions and under conditions of actual use are sufficient in magnitude to warrant serious consideration. Because of the wide range of such variations, electro-acoustic performance data taken by the standard laboratory procedures should be applied with caution to the problem of "fitting" a hearing aid to an individual. The character and magnitude of the possible differences in the electro-acoustic performance of an instrument when it is worn by different persons under various conditions should be borne in mind in the design of the aid. Sufficient gain, power-output capacity, and control of frequency-response characteristics should be provided, if possible, to allow the wearer to adjust the aid for good operation for the great variety of conditions encountered.

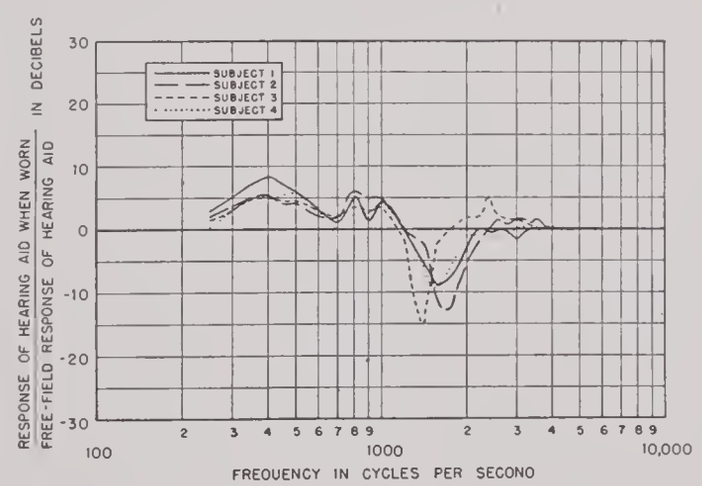


FIGURE 14. Body-baffle correction.

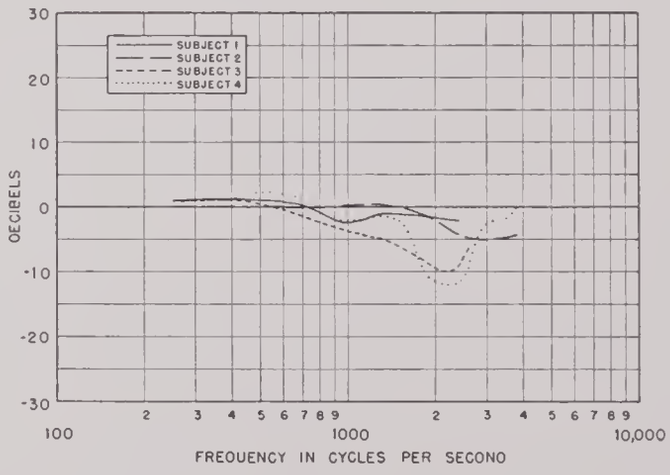


FIGURE 15. Clothing correction.

15.2.2 Psycho-Acoustic Measurements

In evaluating the intelligibility of speech heard via a hearing aid, it would be desirable to use a crew of listeners with impaired hearing, all of whom had the same type and same degree of hearing loss. The difficulty of obtaining such a crew is readily apparent in view of

the wide variation in the severity of naturally occurring or traumatic deafness. For this reason, it was decided at the inception of the psycho-acoustic study to employ a crew of trained listeners with normal hearing and to render them functionally hard-of-hearing during the tests by means of masking noise. The theoretical and experimental bases for this procedure are discussed in reference 1. A flat spectrum of random noise was used to produce a simulated hearing loss of equal severity for all subjects. The amount of loss was easily varied by increasing or decreasing the intensity of the noise. The degree of hearing loss for pure tones occasioned by the various levels of masking noise was measured, but a more general and equally valid expression for the amount of hearing loss was obtained by determining the effective hearing loss in decibels for speech.

The instruments selected for the tests were all of the vacuum-tube type; they were products having the approval, as of January 1944, of the Council of Physical Medicine of the American Medical Association. The instruments were purchased from the stock of local dealers. Where several models of the same make were available, only those models were chosen for test which, according to the dealers' own statements, were most frequently recommended for hard-of-hearing patients.

Each individual model was tested with two or more earphone types when available and with minimum and maximum volume-control settings. In the case of instruments with variable tone control the setting giving the most nearly flat response characteristic and that giving greatest emphasis to the high tones were tested. It was not feasible to study all the possible combinations of the three factors.

The hearing aid to be tested was elastically suspended on a ring stand in an anechoic box, and connected speech was presented to it from a loudspeaker 20 in. distant. For most of the tests the speech used was phonographically recorded. Analysis of the spectrum of this speech signal showed it to be identical with the spectrum obtained when the live voice was substituted for the phonograph and loudspeaker. The earphone of the hearing aid was coupled to a condenser microphone, the output of which was distributed, essentially unchanged, through standard-sized earphones of high fidelity, to four subjects who were seated in a quiet, sound-treated room. The masking noise was introduced electrically into the subjects' earphones through a separate channel. A block diagram of the testing equipment is shown in Figure 16.

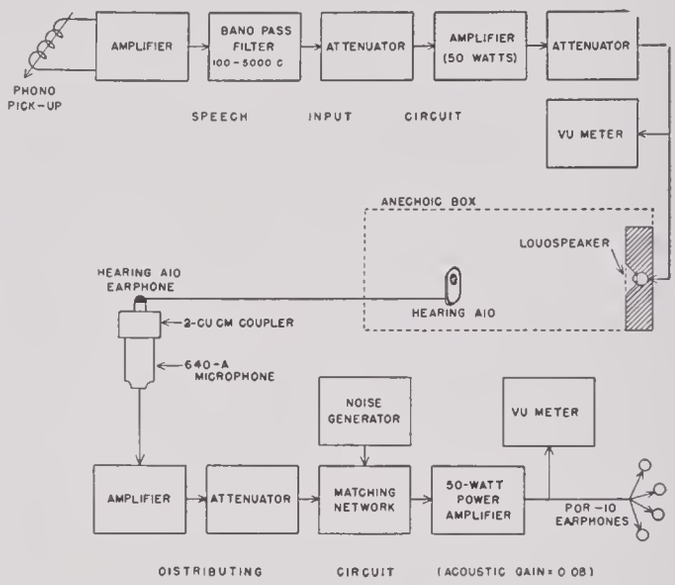


FIGURE 16. Block diagram of the circuits used for testing hearing aids.

One aspect of the performance of the hearing aids was evaluated in terms of the speech-input level at the hearing-aid microphone required to give sufficient output to override various degrees of simulated hearing loss. By plotting the speech-input level required to reach the threshold of intelligibility at each level of the effective hearing loss, a function bounding the area of intelligibility for speech for a given instrument was obtained.[7a] From such a function the minimum speech level which can be made intelligible for a given degree of hearing loss may be read. This speech level is determined primarily by the effective gain of the instrument. The maximum hearing loss which can be overcome at any voice level may also be read from the curve. This value depends primarily on the maximum power output of the hearing aid.

Sample plots of this nature are shown in Figures 17 and 18. The area of intelligibility for speech is bounded above and on the left by the curve representing the threshold of in-

telligibility. The position of this threshold curve varies with the volume-control setting. The upper limit, or "ceiling," is imposed primarily by the maximum power output of the instrument. As a reference, a high-fidelity system was substituted for the hearing aid, and the amplifiers were so adjusted that no acoustic gain was introduced. The area of intelligibility is bounded below and on the right by the threshold for this "no-gain" condition. The effective gain of the hearing aid is then measured, as shown in Figures 17 and 18, as the number of decibels between the threshold-of-intelligibility curve for the reference system and the threshold-of-intelligibility curve obtained when the hearing aid is inserted.

The quality of speech reproduced by the hearing aid was also studied. The listeners were asked to compare the speech transmitted by each instrument with (1) speech of high fidelity and (2) speech which was greatly distorted but still intelligible. A rating on a seven-point scale of quality was assigned to each instrument. These tests were made at a comfortable output level without the use of simulated hearing loss. There is a significant positive correlation (0.62) between the quality ratings and the number of decibels above the limit of linear amplification at which the rating was made. It appears, therefore, that the quality rating depended largely on the degree and character of the limiting and consequent harmonic distortion that was present.

There are countless minor differences between instruments, but all those examined were enough alike to make it doubtful whether any of the untested instruments would provide great advantages based on radical differences in design. For low and medium input levels, output is proportional to input (linear amplification), and the amount of gain is regulated by the gain or volume control. The maximum acoustic output of the hearing aid is definitely limited, usually by the power capacity in the final stage of the amplifier. All instruments introduced a considerable degree of frequency distortion. Below about 500 c and above about 2,500 c, the amplification of most instruments falls sharply. All the powerful instruments were subject to squeal from acoustic feedback if they were used at high gain with a poorly fitting earpiece.

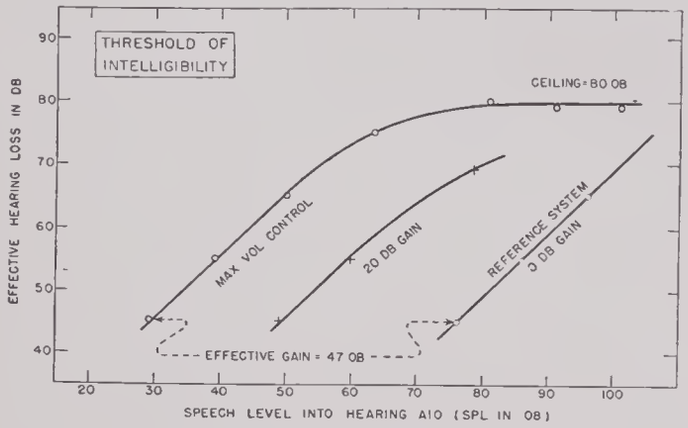


FIGURE 17. Showing threshold of intelligibility at different speech levels with the volume-control setting as parameter. Data for hearing aid X.

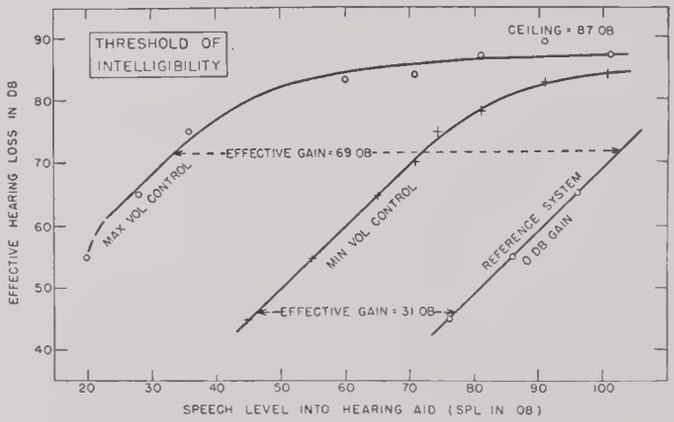


FIGURE 18. Showing threshold of intelligibility at different speech levels with the volume-control setting as parameter. Data for hearing aid Y.

None of the hearing aids, when set for maximum volume, caused painful stimulation of the normal listener's ears, even when the speech-input level was equivalent to 100 db sound-pressure level. Wide differences between instruments are found in terms of the lowest input levels at which speech can be made intelligible. With an effective hearing loss of 75 db, the range of minimum speech levels is from 29 to 80 db for different instruments. The hearing aids vary much less in terms of the maximum effective hearing loss which they are able to overcome at high speech levels. With sufficient speech-input level the weakest instrument is able to give intelligible speech for a hearing loss of 75 db, the most powerful for a hearing loss of 90 db.

15.3 SELECTION OF A HEARING AID

The selection of a hearing aid for any particular patient is a difficult problem because of the variety of requirements for a hearing aid, the variety of existing instruments, and the variety of individual auditory impairments. It is impossible to base the selection of hearing aids on any single or composite "figure of merit" derived from purely physical measurements. No single physical characteristic is sufficiently all-important to be made the basis of any such figure of merit, and it is unlikely that any workable formula for combining the values of several characteristics will be found.

The primary requirement is, of course, that the aid make speech intelligible to the patient. In most cases, however, several different aids will satisfy this requirement equally well, and the problem is to choose among them. At this point several secondary criteria become important to the patient: comfort, weight, expense, reliability, serviceability, quality of the sound, etc. On these matters the patient's personal opinion is the chief criterion.

For many years the principle of "selective amplification" has been widely accepted, implicitly or explicitly, as a guide to the selection of the best hearing aid. The classical argument runs somewhat as follows. A patient's hearing loss is not necessarily uniform as a function of frequency. Many patients become deaf to high-frequency sounds but still hear sounds of low frequency quite well. A smaller number suffer from low-tone deafness, while for a few the tones in the middle of the audible range are most impaired. A very common type of hearing loss is moderate for intermediate frequencies and severe or total for sounds of high pitch. Now the various alternative aids are also known to differ, particularly in respect to their frequency-response characteristics. Some emphasize high tones more than others, some amplify a wider band of frequencies (see Section 15.2.1). It is "obvious," therefore, that the instrument which restores the patient's audiogram most nearly to normal is the best for that patient.

The principle of restoring the audiogram to normal by selective amplification is used as a guide in selecting the particular earphone or tone-control setting, and is made the basis of a specific test, the so-called "aided audiogram," to determine which instrument best fits the patient. The method is often compared to the fitting of eyeglasses to correct defects of vision.

The various objections to the procedure of "audiogram fitting" may be summarized.[10] The audiogram reveals the threshold sensitivity, but listening is usually done well above the threshold level. It has been suggested,[2] accordingly, that it is an equal-loudness contour at the level of comfortable listening that should be fitted rather than the threshold. Unfortunately, it is impractical to determine equal-loudness contours as a routine procedure with untrained patients. The attempt was made at the Deshon General Hospital as a part of the present project, and the determination proved to be either unreliable or unduly time-consuming.

The relative intelligibility obtained with several widely different patterns of frequency response was determined at the Psycho-Acoustic Laboratory for 25 hard-of-hearing ears. Of these, four could be classified as having low-frequency losses, five as having uniform loss at all frequencies, ten showed a gentle slope downward at high frequencies, and six had marked high-frequency losses. This classification of the shapes of the audiograms is admittedly arbitrary, but corresponds roughly to "audiogram-fitting" practices. The frequency responses of the testing equipment included, in addition to the flat system, four "tilts," both rising and falling with frequency at rates of 6 or 12 db per octave (see Figure 19 in Chapter 7). These frequency responses will be referred to as LP-12 (low-pass, sloping at 12 db per octave), LP-6, flat, HP-6 (high-pass, rising at 6 db per octave), and HP-12. Articulation tests were run with each patient for various conditions of frequency distortion, and the results permit the conditions to be rank-ordered for each patient. Two criteria were used as the basis for evaluating the subject's performance on the several frequency patterns.

15.3.1 Maximum Articulation Score

The rank orders obtained on the basis of the

maximum articulation score for each frequency pattern may be summarized as follows:

LP-12. Only two subjects were able to obtain articulation scores higher than 50 per cent and both cases were much more successful with the other conditions. The inferiority of this freqency pattern is clearly indicated.

HP-12. Two of 14 ears did as well with this frequency pattern as with any other, but for at least 10 subjects this pattern was rated as third, fourth, or fifth best.

LP-6. Three out of 22 did as well on this pattern as on any other, but most of the scores (17 cases) ranked the pattern as the third or fourth choice. In no case did the LP-6 give a better score than the flat or the HP-6.

HP-6. Data were available with this frequency pattern for all 25 ears. Twenty-three of these obtained scores which were ranked first, and the remaining two ranked this frequency pattern as second best. However, 18 of the 23 firsts were shared equally with the flat system.

Flat. For 21 of the 25, the score on the flat frequency pattern was assigned the rank of 1. The flat ranked second to the HP-6 in four cases, third in one case.

15.3.2 Maximum Operating Range

The operating range is defined as the range, in decibels, between the lowest speech-input level at which the listener attains an articulation score of 50 per cent and the highest level at which nonlinear distortion again reduces the score to 50 per cent. Nonlinear distortion (peak clipping) was used to protect the listener, and in all cases the distortion began at the same peak intensity (124-db SPL) in the listener's ear.

The maximum articulation scores and the operating range are not independent since they are both derived from the same articulation functions. In general, the two variables are highly correlated, but they may disagree under some circumstances. The degree to which the ranks assigned on the basis of the operating ranges agree with ranks similarly assigned on the basis of the maximum scores can be summarized as follows:

Flat: rankings agree in 19 of 24 cases.
HP-6: rankings agree in 24 of 24 cases.
LP-6: rankings agree in 14 of 21 cases.
HP-12: rankings agree in 8 of 13 cases.
LP-12: rankings agree in 2 of 2 cases.

Thus the agreement is relatively close between these two measures of performance, and the conclusions drawn from maximum articulation scores are supported by the data on the operating ranges. It is clear that the performances of the HP-6 and flat frequency patterns head the list. In every one of the 25 cases either the flat or the HP-6 ranked first. None of the other three patterns competes seriously with these two.

15.3.3 Quality Preferences

For 23 of the 25 ears studied, the listeners made subjective ratings of the quality of the speech. In 14 cases the preferred quality was produced by the same frequency pattern which gave the best performance on the articulation tests, but in 9 instances the quality judgment was either ambiguous or definitely misleading. In some cases, therefore, the patient's personal opinion about a secondary requirement should give way to the primary requirement of intelligible speech.

These results, definitely contrary to the original expectations of the experimenters, seem to show that it is possible to specify the desirable frequency characteristic of a hearing aid more successfully by a simple, general rule than by any interpretation of the patient's audiogram.

15.4 DESIGN OBJECTIVES FOR HEARING AIDS

On the basis of the data obtained on hard-of-hearing subjects, tentative design objectives can be drawn up.[10] Design problems are greatly simplified by the realization that a single, versatile instrument of appropriate general characteristics, with a variable tone control and a semipermanent adjustment of maximum acoustic output can encompass the entire range of requirements.

The overall acoustic frequency characteristics

of a hearing aid should be uniform (without sharp peaks) between 400 c (preferably 300 c) and 3,000 c (preferably 4,000 c). The response contour should be smooth at the maximum power-output level as well as at ordinary operating levels. The high-frequency cutoff can be as abrupt as engineering convenience requires. Below 300 c the frequency response should fall off at a rate of at least 10 db per octave and a sharper cutoff is permissible.

Between the cutoff frequencies of 300 and 4,000 c, the overall slope of the frequency characteristic should be "flat" or should rise toward the high frequencies at a slope of not more than 1 db per octave. An adjustable tone control should provide an alternative slope, rising evenly 6 or 7 db per octave, toward the high frequencies. These two extreme adjustments are required. At least one intermediate setting or a continuously variable tone control is recommended.

Maximum acoustic output should be limited so that the ear is protected against transients. An investigation by the Central Institute for the Deaf[9] determined the thresholds of discomfort, of tickle, and of pain produced by pure tones and by speech for 46 normal and 46 hard-of-hearing ears. The thresholds did not vary as a function of frequency between 300 and 5,600 c. All three thresholds rise systematically with successive test sessions, and both the initial and final threshold in decibels re 0.0002 dyne/cm^2 are given in Table 1. (Figures in parentheses are the limits set by the apparatus, and the median thresholds are above these values by uncertain amounts.) These values give an approximate indication of the permissible upper limit. It will be noted that the normal and hard-of-hearing ears differ little in their tolerable thresholds.

The instantaneous acoustic output, measured as pressure under an earphone at which limiting occurs, should be definitely established, either by a semipermanent adjustment, or by choice of battery voltages, or by the provision of three or four models with significantly different maximum acoustic outputs.

Of the available devices for limiting the peak output level, the simplest is the "peak clipper." Properly adjusted peak clipping protects the ear from discomfort and pain while allowing a predetermined maximum amplitude of signal to be delivered. Compression amplification produces less amplitude distortion than simple peak clipping, and if an effective compressor can be built into a wearable hearing aid it may provide the ideal means of limiting the maximum acoustic output.

TABLE 1. The initial and final thresholds of discomfort, tickle, and pain for normal and hard-of-hearing ears exposed to speech and pure tones (in db re 0.0002 dyne/cm^2).

	Discomfort		Tickle		Pain	
	Initial	Final	Initial	Final	Initial	Final
Normal						
Pure tone	110	123	133	(> 142)	140	(> 142)
Speech	117	130	128	135	138	(> 139)
Hard-of-Hearing						
Pure tone	118	129	129	(> 141)	136	(> 141)
Speech	121	130	129	134	135	(> 137)

The instrument must be sufficiently sensitive and free from electrical or other internally generated noises to allow it to render intelligible to a normal ear the speech delivered to it at a level not more than 10 db above the unaided threshold of intelligibility of that same normal ear. Patients' requirements in regard to amplification vary so widely that it is probably desirable to design at least two or perhaps three different models of hearing aids. For severe hearing losses a maximum gain for speech of 80 db should be provided. For mild cases, an instrument with a properly designed gain control could cover the entire range of desired amplification, but it might be unnecessarily large, heavy, and expensive for the patient who never requires more than 30 or 40 db of acoustic gain. The justification for more

than one model is economy of size and weight and expense, and such considerations may properly determine the maximum gain to be provided by medium-gain and low-gain models. No instrument should, however, have less than 30 db maximum acoustic gain.

The instrument must be provided with an effective gain control, either continuously variable or with numerous intermediate positions between maximum and minimum. This range should be at least 40 db. The gain control must be designed to avoid accidental shift of setting by contact with clothing, and it should operate roughly linearly on a decibel scale.

There should be no electric feedback (squeal) when the instrument is used at maximum gain setting. Acoustic feedback should be dependent on the fit of the earpiece and not on the leakage from the receiver itself.

According to the experimental evidence, such an instrument would provide the maximum possible assistance to all those patients for whom hearing aids are prescribed.

15.5 WORK DONE AT THE CENTRAL INSTITUTE FOR THE DEAF

The Central Institute for the Deaf, St. Louis, Mo., participated in the Hearing Aid Project under Contract OEMsr-1201. Its chief contributions were:

1. An exhaustive study of the thresholds of tolerance, discomfort, tickle, and pain.
2. The assembly and installation at Army and Navy Aural Rehabilitation Centers of electro-acoustic equipment for clinical auditory tests.

15.5.1 Tests of Tolerance

The thresholds of *discomfort*, of *tickle*, and of *pain* produced by pure tones and by speech were determined in over 11,000 observations, on 46 normal and 46 hard-of-hearing ears.

The initial thresholds for pure tones of pain and of tickle lie at about 140 db and 133 db, respectively, for all frequencies from 250 to 5,600 c. The median intensities are greater for the normal group than for the hard-of-hearing group, owing to the presence of more "tender" hard-of-hearing individuals reporting tickle and sometimes pain between 120 db and 130 db.

The threshold of discomfort approximates an equal-loudness contour and shows a broad minimum (3 db below the mean) between 1,400 and 4,000 c. The initial median threshold for the normal (110 db) lies below the threshold for the hard-of-hearing (120 db), but there is great dispersion, particularly among the hard-of-hearing.

Similar determinations made with carefully monitored (recorded) speech gave a similar set of values (see Section 15.4). Tickle is a more constant phenomenon for speech than for pure tones, and has a lower threshold as measured by a VU meter; but pure tones cause discomfort at lower threshold in normal ears than does speech.

All three thresholds rise systematically and significantly with successive test sessions either daily or weekly, and approach limiting values after several test sessions. The initial and the final values for the various thresholds have been presented in Section 15.4, Table 1.

The increased tolerance is largely but not entirely retained after an interval of a week, and about half of the increase is retained for at least 20 weeks without additional exposures.

Development of tolerance for speech elevates slightly the tolerance thresholds for pure tones. Pure tones are less effective in elevating the tolerance thresholds for speech. Development of tolerance for speech or for pure tones in one ear does not increase the corresponding tolerance of the other ear of the same individual.

Tolerance may be developed effectively by exposure to loud speech at a level just below the threshold of discomfort for several minutes a day on three or four successive days.

15.5.2 Electro-Acoustic Equipment

Electro-acoustic equipment for clinical auditory tests was installed at Borden General Hospital, Chickasha, Oklahoma, at Hoff General Hospital, Santa Barbara, California, and at the U. S. Naval Hospital, Philadelphia, Pennsyl-

vania. The details of the equipment in the several installations varied slightly as improvements were introduced on the basis of experience and the increasing recognition of the importance of free-field tests based on recorded or live speech.

All assemblies were basically two-channel systems of high fidelity, one to deliver the test material to the subject in a suitably sound-treated test chamber and the other a monitoring channel for measuring sound levels in the chamber and for talkback from the subject. Oscillators, turntables, microphones, and special noise generators served as sources for the test sounds. An amplifier provided the test channel with 120 db of gain with undistorted power output. A loudspeaker with a frequency response reasonably flat from 100 to 9,000 c delivered a maximum output of 115 db at a distance of 2 m on its axis. Appropriate provision was made for mixing various input materials and regulating their intensities accurately at known levels.

With these installations it was possible to conduct all the various tests, except routine clinical audiometry, suggested by other sections of the Hearing Aid Project, both for assessing the impairment of hearing and for the selection of hearing aids. The possible tests included:

A. Monaural tests of the patient without a hearing aid. The apparatus is used essentially as an audiometer.
   1. Thresholds for pure tones.
   2. Thresholds for speech.
   3. Tolerance, i.e., pain or discomfort thresholds, for pure tones.
   4. Tolerance for speech.
   5. Articulation tests with suitable word lists.

B. Free-field binaural tests without a hearing aid. The same tests listed for monaural study may be performed under binaural free-field conditions.

C. Free-field tests with hearing aid.
   1. Pure-tone aided thresholds.
   2. Aided threshold for speech.
   3. Tolerance for loud speech with hearing aid.
   4. Articulation tests.
   5. Judgments of quality.
   6. Masking and annoyance effects of various background noises on speech thresholds or articulation scores.

Chapter 16

SONIC POSITIONING DEVICES AND DIRECTION FINDING

THE USE of sound signals for direction finding and positioning for military purposes has been suggested many times. These suggestions were apparently mostly prompted by reference to the everyday experience of easy and successful aural localization rather than by considerations of the physical problems involved. In reality, the problem is a difficult one which admits of practicable solutions only in certain special cases and only after considerable difficulties have been overcome.

Since World War I *underwater* sonic direction-finding devices have been used with success. The following discussion deals with certain problems of sonic direction finding in *air*. The nature of the subject matter and the relative scarcity of relevant information will justify the inclusion of some material which may be of general interest but is not directly related to the final solution.

16.1 SONIC DIRECTION FINDING FOR AIRCRAFT INTERCEPTION

One of the first problems submitted to the Electro-Acoustic Laboratory by the Services was the investigation of the possibility of using an acoustic direction-finding device for blind interception of aircraft. The device would be used by an attacking plane which has been able, by other means of detection, to take up a position a relatively short distance behind the enemy plane. The device would then enable the plane to close in on the enemy and down it by gunfire.

A survey of the problem was made and the (largely negative) findings were published along with exploratory experimental data in a report form.[1] The method of attack proposed therein is the use of a highly directional array of microphones suitably mounted on the interceptor. The acoustic energy generated by the propellers, engine exhausts, etc., of the target plane and radiated toward the rear is picked up by the microphone array. The electric signal is presented to the attacking pilot in the form of a suitably directional visual indication giving his position relative to the source of sound on the enemy plane. Clearly, it is all-important to secure a signal-to-noise ratio at the receiving microphone array high enough for reliable operation of the indicating device. The report mentioned above concludes that this is not possible, mainly for the following reasons.

1. Propellers radiate very little sound in a direction opposite to the direction of flight, especially at medium and high frequencies.
2. The noise at the microphone location due to the interceptor's propellers and air turbulence near the microphones is predominant.
3. The design of a microphone capable of operation in high wind velocities and suitable for mounting on aircraft is a problem of considerable difficulty.

16.2 SONIC POSITIONING DEVICE FOR USE ON TOWED GLIDERS

For obvious reasons, it is necessary for towed gliders to maintain in flight, within certain limits, a predetermined position relative to the tow plane. When visibility is good, the glider pilot is able to accomplish this simply by watching the tow plane and piloting accordingly. But under conditions of poor visibility, some other means must be at his disposal to determine his position correctly.

Of the many possible approaches, only acoustic methods will be considered here. Work on a *sonic positioning indicator* [SPI] was undertaken by the Electro-Acoustic Laboratory in 1942 at the request of the Army Air Forces. The project resulted in the development of a lightweight model suitable for serving as preproduction sample, which was flight tested late in 1943. In the opinion of the witnessing laboratory personnel, the tests were successful, although a series of minor revisions appeared to be necessary. These tests are described in detail, along with a discussion of the theory

and operation of the SPI, in a report issued in February 1944.[2] Several other methods of attack are also discussed there.

From comparison with the problem discussed in Section 16.1, it appears that success was in no small measure due to four circumstances inherent in the problem. First, the tow plane is friendly and a sound source can be installed to provide adequate acoustic signal level at optimum frequency at the glider. Second, there is no interfering propeller noise at the glider; hence a wider choice of microphone locations and more favorable signal-to-noise ratios are possible. Third, the cruising airspeeds of towed

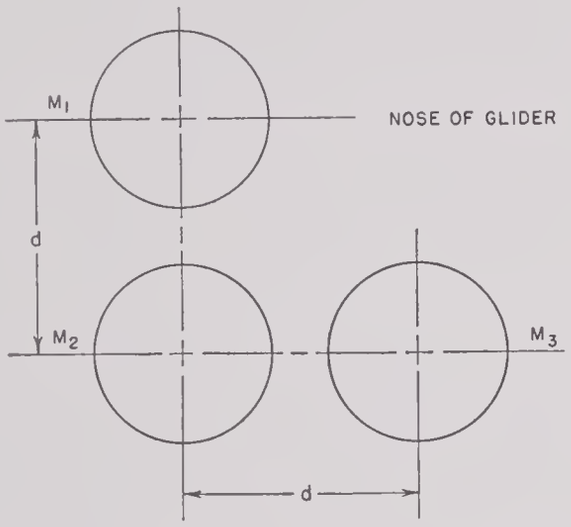


FIGURE 1. Location of microphones in glider nose.

gliders are at present rarely, if ever, above 200 mph. Hence, design of a microphone capable of operation at such low airspeeds seemed feasible. Fourth, the length of the tow rope is only about 300-400 ft.

The SPI operates as follows: A *source of sound* is mounted on the tow plane such that most of its radiated acoustic energy is directed backward toward the glider. Three *microphones* mounted on the nose of the glider, as shown in Figure 1, convert some of this acoustic energy into electric energy. The electric signal is transmitted to the *sonic receiver* [SR] mounted in the glider which furnishes suitable visual indications to the glider pilot. The microphones are positioned in such a way that when the glider is directly behind the tow plane sound leaving the source on the tow plane at a particular instant strikes the three microphones simultaneously, and the acoustic signals reaching the three microphones are in phase.

If, however, the glider moves to some other position not directly behind the tow plane, the signal, in general, no longer reaches the three microphones at the same instant and hence the three signals are no longer in phase. This is illustrated in Figure 2 in which, for reasons of simplicity, only two microphones are shown. It is evident that the signal reaching microphone M_2 travels a distance farther than the signal reaching M_3. The phase difference between the two signals is

$$\phi_h = 2\pi \frac{x}{\lambda} \tag{1}$$

where ϕ_h is the phase difference, in radians, due to an angle Θ_h measured in a horizontal

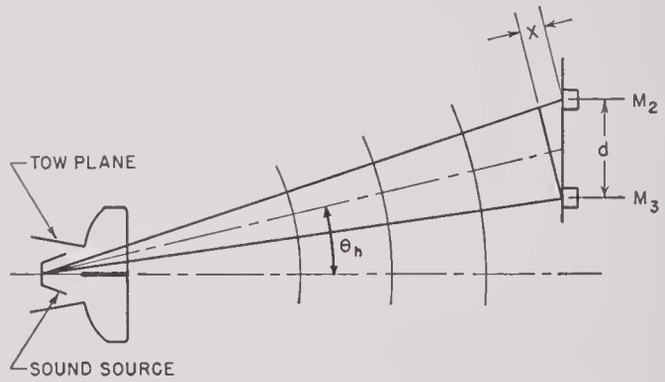


FIGURE 2. Showing the phase difference between the signals reaching microphones M_1 and M_2.

plane and λ is the wavelength of the sound signal. But

$$x = d \sin \Theta_h, \tag{2}$$

where d is the separation of M_2 and M_3. Hence

$$\phi_h = \frac{2\pi d}{\lambda} \sin \Theta_h. \tag{3}$$

Since for most angles of interest, $\sin \Theta_h \approx \Theta_h$,

$$\phi_h \approx \frac{2\pi d}{\lambda} \Theta_h. \tag{4}$$

Similarly, for an angular displacement of the glider measured in a vertical plane the phase

difference ϕ_v between the signals reaching microphones M_1 and M_2 is given by

$$\phi_v \approx \frac{2\pi d}{\lambda} \Theta_v. \tag{5}$$

Hence ϕ_v is a direct measure of the glider position in a vertical plane (above or below the tow plane), while ϕ_h is a measure of the glider position in a horizontal plane (to the left or right of the tow plane). It is the function of the SR, the electronic equipment to which the microphones are connected, to convert these

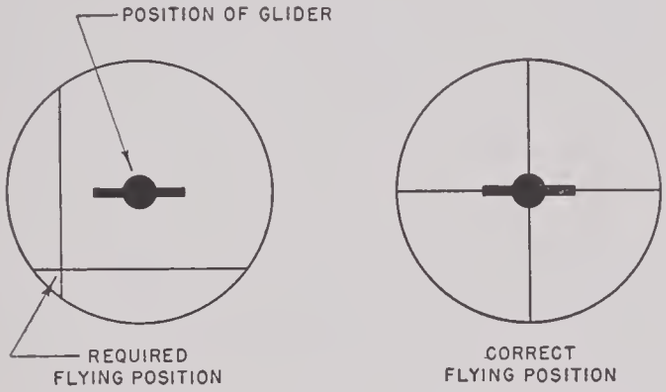


FIGURE 3. Indication of glider position with respect to tow plane.

two phase differences into readings of a meter by means of phase-detecting circuits. The meter is of the crossed-pointer type which indicates to the glider pilot his actual position relative to the desired one. If the glider is, say, above and to the right of the correct position, the meter needles take the position shown in Figure 3.

It has been tacitly assumed so far that the correct flying position for the glider is directly behind the tow plane. In reality, glider pilots prefer to fly somewhat above the tow plane for single tows. In case of multiple tows, they must fly at appreciable values of the angle Θ_h. Controls, therefore, are provided to preset the meter zero accordingly.

16.2.1 Aerodynamic Noise at Type CG-4A Gliders

The choice of microphone position at the glider and the choice of the operating frequency of 4,000 c for the SPI was a result of a thorough study of the noise produced by air turbulence at various positions at or near the surface of Type CG-4A gliders. A number of calibrated experimental microphones in streamlined housings were mounted at various positions on the glider. The noise levels at these microphones were measured for various airspeeds by means of a sound analyzer in bands 200 c wide.

Typical values of noise levels for a microphone mounted on the glider nose are given in Table 1 as a function of frequency.

TABLE 1. Ambient noise levels at nose of CG-4A glider. (Bandwidth 200 c; indicated airspeed 135 mph; altitude 5,700 ft.)

Frequency (cycles per second)	Noise level (db re 0.0002 dyne/cm²)
200	120.5
400	112
600	109.5
800	110.5
1,000	107
1,400	93
1,800	81.5
2,200	77.5
2,600	75.5
2,800	75
3,200	73
3,600	70.5
4,000	71

Inspection of the values in Table 1 shows that the low-frequency noise is predominant. The levels are comparatively low at frequencies beyond about 2,000 c. It is in this frequency range, then, that the transmission frequency should be located for favorable signal-to-noise ratios. This assumes, of course, the feasibility of producing a given signal level at the glider. Some of these questions will be taken up later.

From inspection of similar data for different microphone locations it appeared that at low frequencies the noise levels are greatest at the nose of the glider and least out on the wings at locations sufficiently removed from the wing profile. At frequencies above about 2,000 c, however, the reverse is true in some instances. For this and reasons of practicability, the glider nose has been chosen as microphone location.

16.2.2 Transmission of Acoustic Signals from Tow Plane to Glider

When the tow plane and glider are moving through the air, the air through which the

sound signal must pass is in a turbulent state. As a result of this and other factors, the density of the air through which the signal is transmitted varies irregularly with both time and position. Hence, the signal reaching the microphones does not have a constant level, but fluctuates sometimes by large amounts.

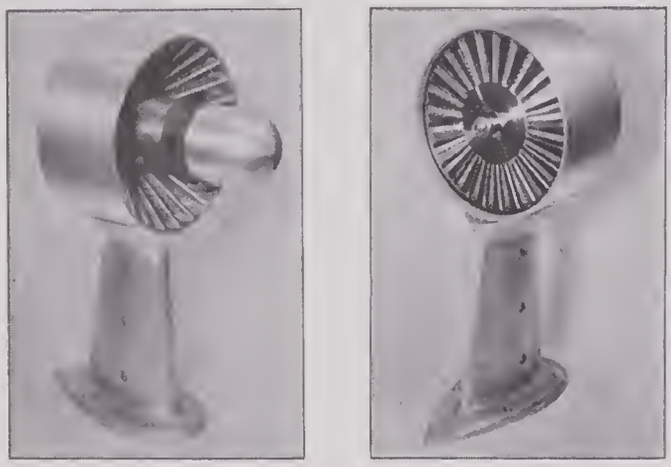


FIGURE 4. Wind-driven siren.

The following values of rapid (about 10 msec minimum duration) fluctuations of the signal were recorded in flight using a certain experimental sound source: about 10 db at 500 c, 15 db at 2,000 c, 20 db at 4,000 c, and 25 db at about 7,000 c.

Turbulence of the air intervening between tow plane and glider causes distortion of the wave fronts of the progressive sound wave emitted by the sound source on the tow plane. This would naturally affect the operation of the sonic receiver. No direct measurements of these distortions of the wave front reaching the glider were made. Preliminary flight tests of an SPI showed, however, that reliable operation of the SR and phase-discriminating circuits could be maintained, provided a favorable average signal-to-noise ratio could be obtained at the glider. This could be realized by using a sound source which was capable of generating a sound level at the glider of about 85 db (re 0.0002 dyne/cm^2) at a frequency of 4,000 c. Band-pass filters of 200-c bandwidth were used in each of the three microphone channels.

This choice of signal frequency was a compromise between favorable signal-to-noise ratio and practicable physical dimensions and power requirements of the sound source on one hand and signal fluctuations at the glider and atmospheric scattering and absorption on the other.

FIGURE 5. Sound source mounted on tow plane. (Army Air Forces photo.)

16.2.3

The Sound Source

It consists of a wind-driven siren mounted on the tow plane and aimed toward the glider

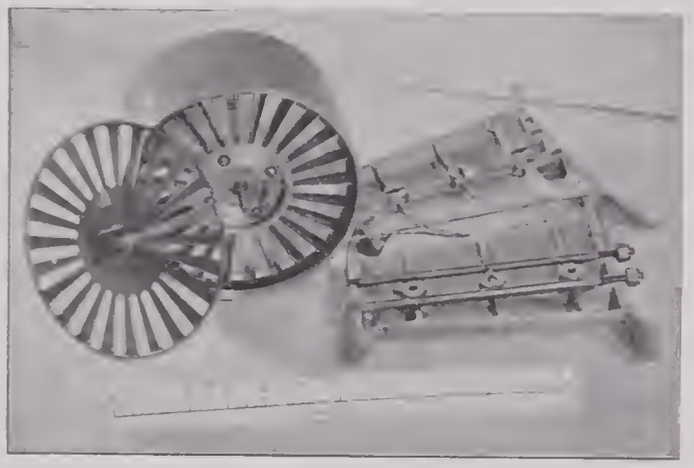


FIGURE 6. Siren with rotor detached.

(see Figures 4 and 5). The siren was adopted after thorough trials of a great number of other devices, ranging from organ pipes to modulated air-flow loudspeakers.[2a]

Some of the details of the construction of the siren may be gleaned from Figures 6 and 7. The cylindrical housing encloses 24 radial fans. They form long passages through which the air entering at one end is forced in flight. At the other end of the cylinder these passages are alternately opened and closed by a slotted rotor disk, whose slots correspond to the slots in the housing. It is driven by the air flow at 10,000 rpm, interrupting it 4,000 times per second. For frequency-control purposes, a small electric control motor with a centrifugally operated switch is provided on the shaft. This control motor is essential to keep the frequency of the acoustic signal sufficiently constant to fall within the pass band of the filters in the SR. The shape of the cross section of the rotor blades and the spacing between rotor and stator are critical design parameters for a given model. The width of the radiation beam of the siren is about ± 20 degrees for points 6 db down from the value on the axis. The total weight of the sonic equipment installed in the tow plane can be kept below about 11 lb without difficulty.

16.2.4

The Sonic Receiver

A block diagram of the SR is shown in Figure 8. The three microphones supply the electric signals to three identical channels, consisting of band-pass filters and amplifiers with suitable automatic volume control. The phase

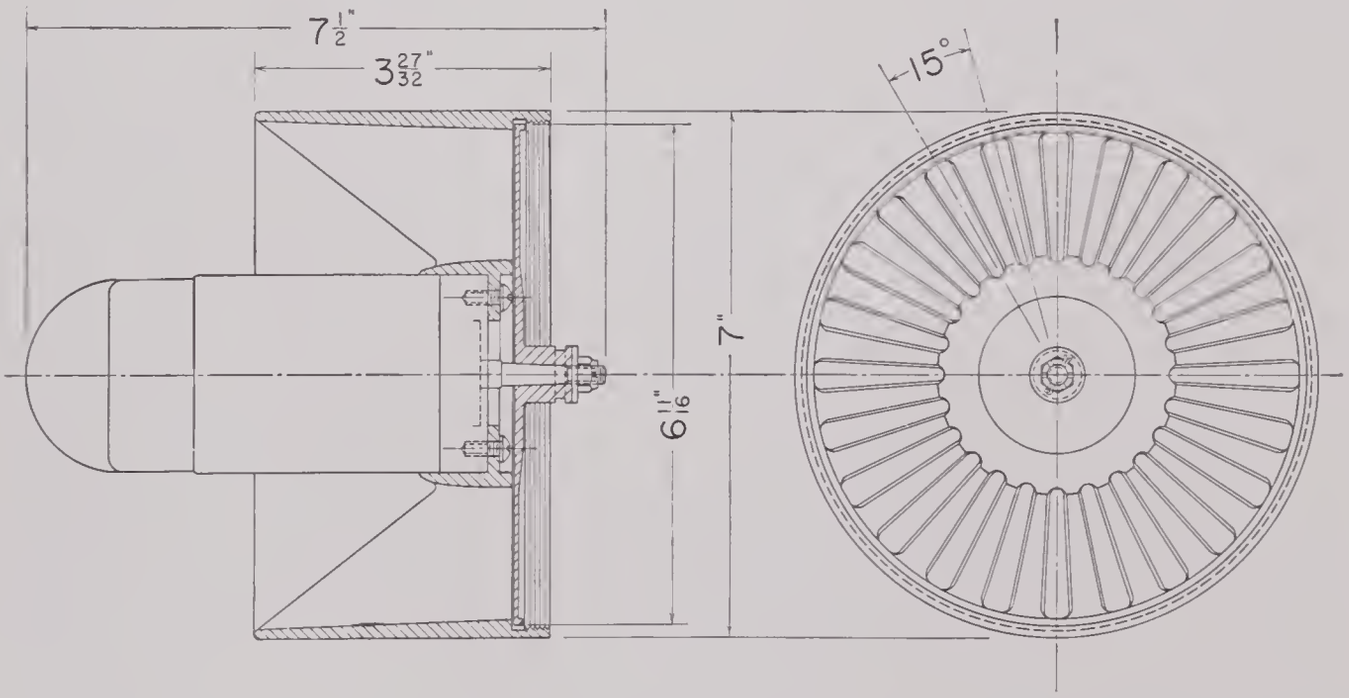


FIGURE 7. Cross section of siren.

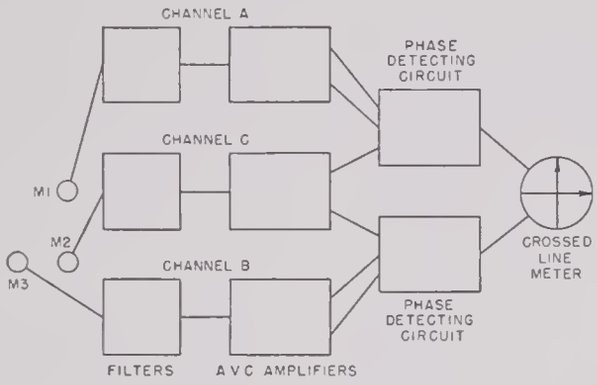


FIGURE 8. Diagram of sonic receiver.

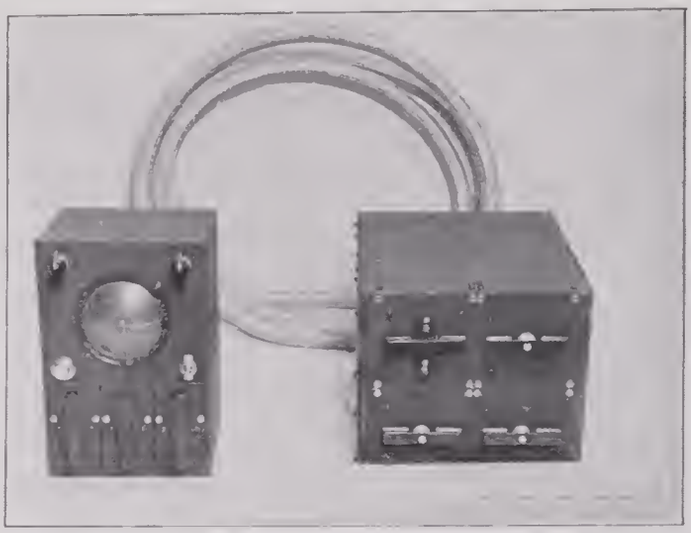


FIGURE 9. Sonic receiver.

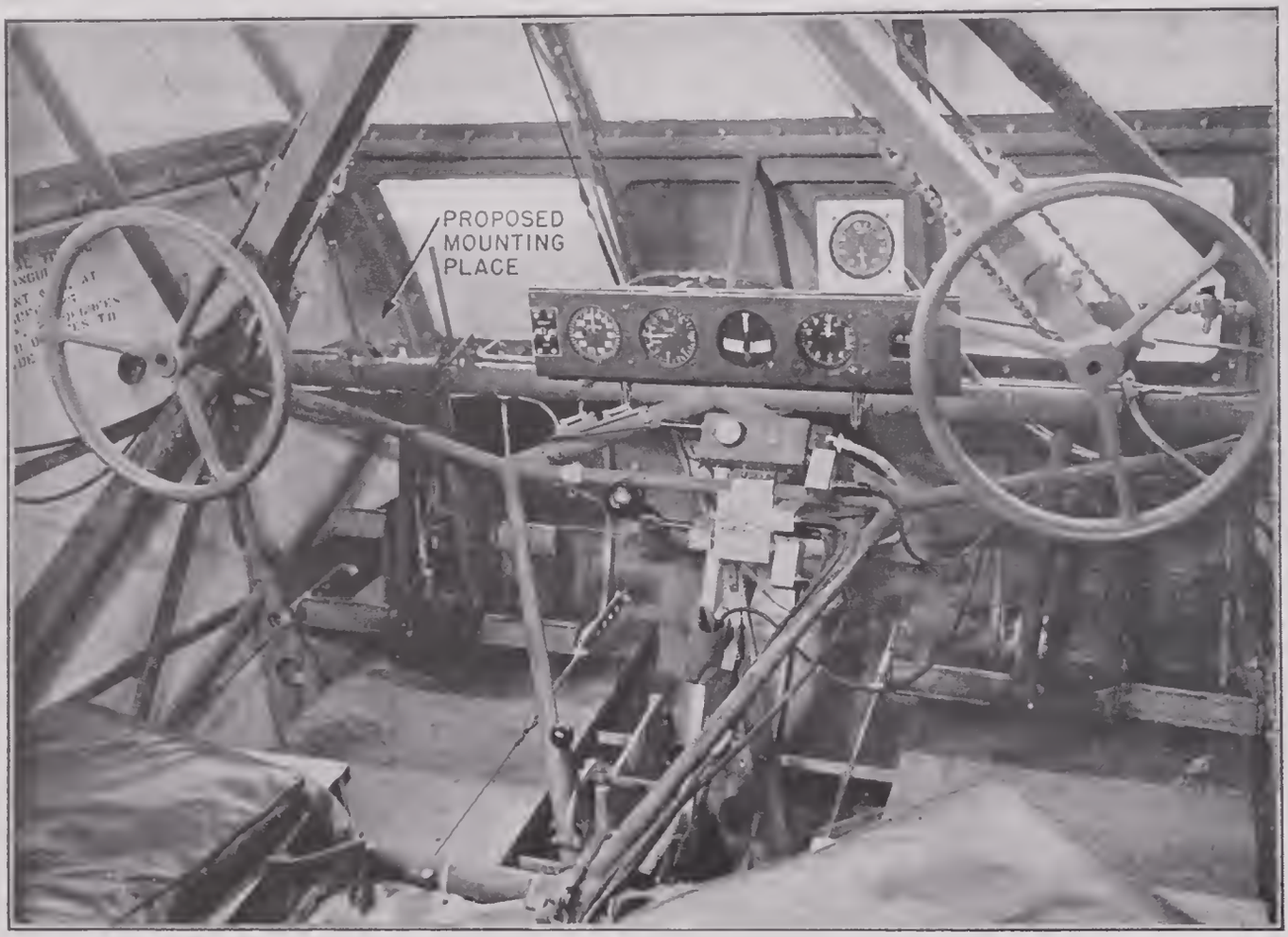


FIGURE 10. Proposed location of sonic receiver in CG-4A glider. (Army Air Forces photo.)

differences between the microphones M_1 and M_2, and M_3 and M_2, respectively, are transformed into deflections of the indicating meter by two phase-detecting circuits of conventional type.

The preproduction model of the SR is shown in Figure 9. It consists of two units: the control box (containing the position-indicating meter

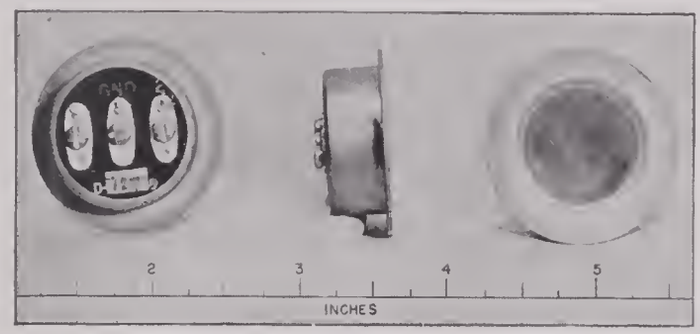


FIGURE 11. Microphone unit used on glider.

and the filters) and the amplifier box (containing the amplifiers and the power-supply dynamotor). The two units are connected by means of the shielded cable shown in the figure. Figure 10 shows the proposed mounting place for the control box for the SR near the pilot's position in a CG-4A type glider.

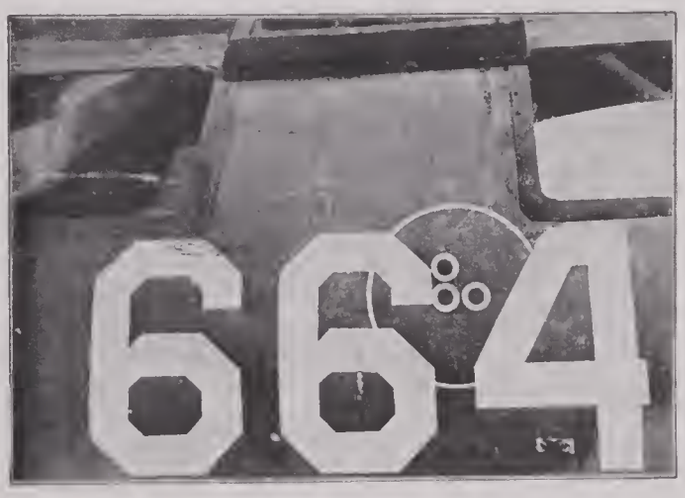


FIGURE 12. Location of microphones on CG-4A glider. (Army Air Forces photo.)

One of the three microphones is shown in Figure 11. They are modified Western Electric Type MC-253 magnetic units. The design of the case has been chosen so that the microphones may be mounted in holes in the glider nose (see Figure 12) flush with the surface of the glider. The microphones are operated electrically in an "off-resonance" condition which tends to minimize matching difficulties and change in microphone phase characteristics, resulting from changes in the conditions of ambient temperature, shocks, etc. A sample frequency-response curve is plotted in Figure

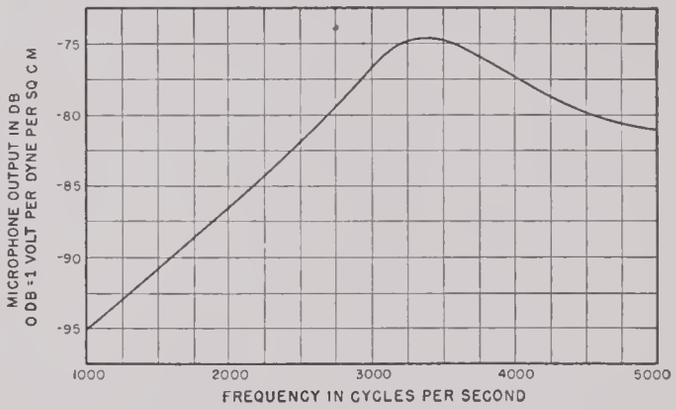


FIGURE 13. Frequency-response characteristic of microphones.

13. Graphs indicating meter current versus position angle are shown in Figure 14 for input signals of varying quality. These tests were made in the laboratory under simulated flight conditions.[2b] The accuracy of the device

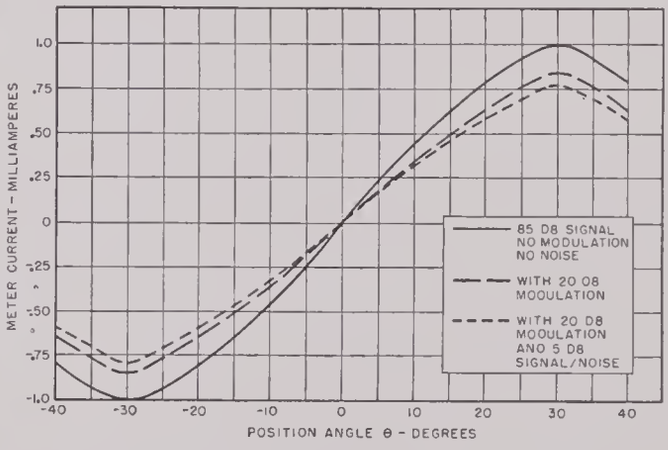


FIGURE 14. Showing meter current vs position angle for signals of varying quality.

seems to be satisfactory for practical purposes.

The entire equipment installed in the glider weighs about 30 lb. The equipment draws about 4.5 amp from the 12-v glider battery.

The device was not adopted by the U. S. Army Air Forces.

Chapter 17

SONIC TRUE AIRSPEED INDICATOR

17.1 INTRODUCTION

THE WORK REPORTED in this chapter was begun early in 1944 by the Electro-Acoustic Laboratory at the request of the Navy. A letter from the Assistant Coordinator of Research and Development to the National Defense Research Committee [NDRC] reads, in part, as follows: "The Bureau of Aeronautics desires the development of a sonic true airspeed indicator. This device . . . depends on the measurement of the difference in time taken by a sound impulse to travel between two points on the aircraft, when traveling with and against the relative air flow."

Airspeed indicators developed in the past depend for their operation on a differential pressure effect, the cooling effect of an air stream or the deflection of a rotatable member. These indicators include the pitot-static system, the Venturi tube, the hot wire anemometer, the deflecting plate, the cup anemometer, and the vane anemometer. A survey of the conventional types of airspeed-measuring devices may be found in a report.[2]

At the present time, the pitot-static system is by far the most widely used type of airspeed indicator. This device measures the difference between the pressure developed by the impact of the air stream on the front of a tube (pitot pressure) and the pressure in a certain region where stable air flow exists (static pressure). Since the impact pressure is proportional to the air density, which, in turn, is a function of altitude and temperature, the pitot-static indicator does not yield true airspeed directly. In general, corrections have to be applied for temperature and ambient static pressure to obtain true airspeed indications. These corrections can be made manually, point by point, or with the aid of specially designed computers in a continuous manner.

The need for accurate and continuous true airspeed indications arises primarily in connection with ground-position indicating devices and devices measuring the distance traveled relative to the air, for use in fast high-altitude, long-range aircraft. True airspeed is defined as the speed of the aircraft relative to the air at a large distance. Hence it is evident that all airspeed indicators are subject to the so-called installation error (in addition to any other errors), since they measure the speed of the aircraft relative to the air in its vicinity. This air is, in general, not stationary with respect to the air at a large distance because of the disturbances set up in flight.

Some of the merits and demerits of the pitot-static system can be stated, in brief, as follows:

1. Corrections for temperature and pressure must be applied. Special computing mechanisms are necessary. Accurate and meaningful measurements of temperature and pressure are difficult in many instances.

2. Pitot-static indicators begin to fail when the stream velocity reaches about eight-tenths of the local sonic velocity. They cannot be used at supersonic speeds.

3. The overall error of pitot-static indicators is least at medium airspeeds. It becomes larger both at low and at high stream velocities.

4. The pitot tube and its mountings disturb the air stream by their presence.

5. The pitot tube is simple, light, rugged, and easy to duplicate.

On the other hand, the possible advantages of a *sonic true airspeed indicator* [STASI] which one might hope to realize are:

1. Except for installation error, true airspeed is measured.

2. Since the velocity of sound is virtually independent of pressure, the indications of the STASI are independent of altitude.

3. It is possible to arrange the acoustic system of the STASI in such a way that the indications are also independent of temperature.

4. On aircraft and in wind tunnels, the sound source and microphones can be mounted flush in an airfoil or wind tunnel wall. Hence stream disturbances are practically nonexistent.

5. Operation is possible at high airspeeds, including operation at supersonic speeds.

6. Continuous true airspeed indications can be secured with comparative ease.

On the other hand, the working accuracy is as yet undetermined and may be larger than for the pitot-static system. The STASI is of necessity more complex than the pitot-static system.

At the present stage of the art the following applications for a STASI can be listed, in the order in which they are feasible and realizable at the present time:

1. Continuous true airspeed indications at low airspeeds (order of 100 mph). Present pitot-static installations on lighter-than-air ships are unsatisfactory and can with advantage be replaced by STASI installations.

2. Calibration of wind tunnels. A method has been worked out by which the velocity distribution in a wind tunnel can be determined by sonic means without appreciably disturbing the stream.

17.2 GENERAL THEORY OF OPERATION

The measurement of airspeed by sonic methods depends on the fact that the transit time of a sound wave between two fixed points in an air stream is a function of the airspeed in the region between the points. Analytical expressions of the relation between transit time and airspeed can easily be obtained for the case of constant sonic velocity and airspeed.[4] It is assumed here that the acoustic signal is a continuous wave having a single frequency. This offers certain practical advantages over devices using sound pulses.

Two simple arrangements of the basic sonic array which consists of a sound *source* (S) and two *microphones* or *receivers* (R_1, R_2) suggest themselves: one where R_1, S, and R_2 are aligned in the order given and the other where the sound source and one of the microphones are interchanged. Measurements of the phase angles between the electric signals generated by the microphones or of the phase angles between the microphone signals and the electric input to the source yield the desired data for computation of the component of true airspeed in the direction of the line of the array. In Table 1 a summary of the operational characteristics of sonic arrays is given. From inspection of frequency and temperature errors it can be seen that a (symmetrical) array can be built which is free from temperature error. It is this type of array which was flight-tested in a Navy airship as described in Section 17.4.

17.3 EXPERIMENTAL STUDY OF SOUND PROPAGATION IN MOVING AIR

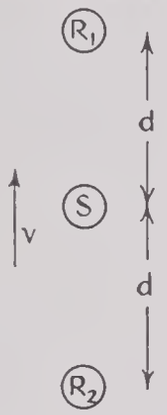
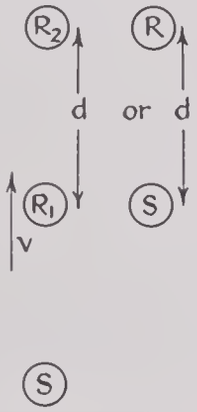
17.3.1 Wind Tunnel Studies at Low Airspeeds

In the discussion of the preceding section it was tacitly assumed that the acoustic arrays in their practical embodiments were mounted flush in an airfoil or some other surface bounding the flow. It becomes all-important, therefore, to study the propagation of sound along boundaries of an air stream, especially at high stream velocities. There is no a priori assurance that a sound signal would propagate along such boundaries as it would in free space. Normal propagation was assumed in deriving the expressions in Table 1.

In order to explore the conditions existing in the boundary layer an extensive test program was undertaken at the Electro-Acoustic Laboratory and at the Gordon McKay Laboratory of the Graduate School of Engineering at Harvard University.

As a first step, an acoustic array of the asymmetrical type was tested, mounted in the floor of the Harvard low-speed wind tunnel. A conventional electronic phasemeter was used to determine the phase differences at an operating frequency of 4,000 c. The microphones and the sound source were mounted in a streamlined wooden "fish" which projected up from the tunnel floor about 2 in. A housing of sound-absorbing material was placed over the array to minimize troublesome acoustic reflections. The airspeeds as computed from the measurements of phase angle were compared with the values obtained from pitot-static measure-

TABLE 1. Summary of operational characteristics of various sonic airspeed measuring arrays.

Array	Airspeed	Quantity Measured	Frequency Error	Temperature Error
	$v = \frac{d}{2}\left(\frac{1}{t_1} - \frac{1}{t_2}\right)$	$\left(\frac{1}{t_1} - \frac{1}{t_2}\right)$	none	none
	$v = \pi f d\left(\frac{1}{\phi_1} - \frac{1}{\phi_2}\right)$	$\left(\frac{1}{\phi_1} - \frac{1}{\phi_2}\right)$	$\frac{v_{meas.}}{v_{true}} \propto \frac{1}{f}$	none
same as above	$v = \frac{c^2 t}{2d}\left(1 - \frac{v}{c^2}\right)$	t	none	$\frac{v_{meas.}}{v_{true}} \doteq \frac{T_0}{T}\left(1 - \frac{v^2_{meas.}}{c_0^2}\left(1 - \frac{T}{T_0}\right)\right)$
	$v = \frac{c^2 \phi}{4\pi f d}\left(1 - \frac{v^2}{c^2}\right)$	ϕ	$\frac{v_{meas.}}{v_{true}} \propto f$	same as above
	$v = \frac{d}{t} - c$	t	none	$v_{meas.} - v_{true} = c_0\left(\sqrt{\frac{T}{T_0}} - 1\right)$
S	$v = \frac{2\pi f d}{\phi} - c$	ϕ	$v_{meas.} - v_{true} = (c + v_{meas.})\left(1 - \frac{f}{f_0}\right)$	same as above

c Sonic velocity at temperature T

c_0 Sonic velocity at temperature T_0

d Separation between microphone and source

v Velocity of air stream

f Frequency of signal

t_i Transit time of signal between source S and microphone R_i $(i = 1,2)$

ϕi Phase angle between the source voltage and the microphone output $(i = 1,2)$

$\phi = \phi_2 - \phi_1$ Phase angle between the two microphone outputs

T Absolute temperature

ments. Table 2 gives the results of this comparison.

TABLE 2. Results of airspeed measurements in a low-speed wind tunnel by the sonic method.

V Sonic (mph)	V Pitot (mph)	Difference (per cent)
62.1	58.8	+5.5
85.3	86.2	—1.2
101.2	104.0	—2.7
111.1	120.0	—7.4
122.3	134.0	—8.7

It was concluded that (a) the STASI is a workable device at low airspeeds, (b) there are indications of systematic deviations of the airspeed obtained by the sonic method from the value obtained by the pitot-static method. These deviations tend to increase with airspeed.

17.3.2 Studies in a High-Speed Wind Tunnel: Methods and Apparatus

In order to explore the nature of these deviations further and to attack the whole problem from a broader angle, an experimental study of sound propagation in moving air and an investigation of the theory of such propagation was initiated, using the Harvard high-speed wind tunnel. This tunnel is of the suction type with one end open to the atmosphere. Mach numbers[a] (M) as high as 1.8 can be reached. The test section is 1¾ in. wide and has a height variable between 4 and 6 in. Optically flat glass sidewalls 8 in. square can be placed anywhere along a 2-ft section of the tunnel.

[a] The Mach number M is defined here as the ratio of the stream velocity to the local sonic velocity.

Thus the flow can be observed optically by the striation, or *schlieren*, method. Briefly, this method is a way of making visible variations in the index of refraction of the air in the tunnel. In the present case, these variations are produced by an acoustic compression wave generated by the discharge of a spark gap located in the air stream. By photographing these compression waves at successive known time intervals by means of a brilliant light flash of extremely short duration successive snapshots of the compression wave are obtained. From the shape and structure of these wave fronts, conclusions can be drawn with reference to the propagation of sound waves in moving air.

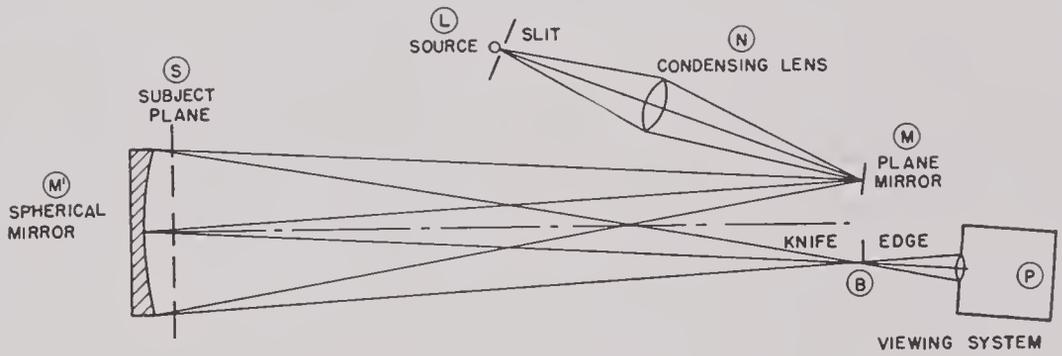


FIGURE 1. Striation coincidence system.

Of the many possible arrangements of the optical elements in a striation apparatus the so-called coincidence system was used. Such a system is shown schematically in Figure 1. The light source is a small, sharp-edged mirror M upon which an image of the source of light L is formed by lens N. M is located near the center of curvature of the large spherical mirror M' but slightly off the axis. The subject plane S of interest is immediately in front of M'. After traversing the subject plane *twice* the light is partially intercepted by the knife edge B located at the conjugate point opposite the source image on mirror M. A camera or ground-glass screen can be placed at P to allow photographic or visual observation. This system can be made free of all optical aberrations with relative ease.

If the air in the subject plane S is of uniform index of refraction, the field of view at P will be uniformly illuminated. If, however, there

is a local change in index of refraction (schliere or striation) in the subject plane, due, say, to a sound wave, a change in the illumination at the corresponding position in the image plane P will be apparent. Note that this arrangement is sensitive only to striation causing deflections of light perpendicular to the knife edge.

In the experiments to be described, the wind tunnel with its glass walls was located in the subject plane S as indicated in Figure 2. To obtain snapshots of sound waves generated by sparks in the wind tunnel electronic equipment is used, a block diagram of which is shown in Figure 3. The chain of operations is as follows: A push button initiates the discharge of the

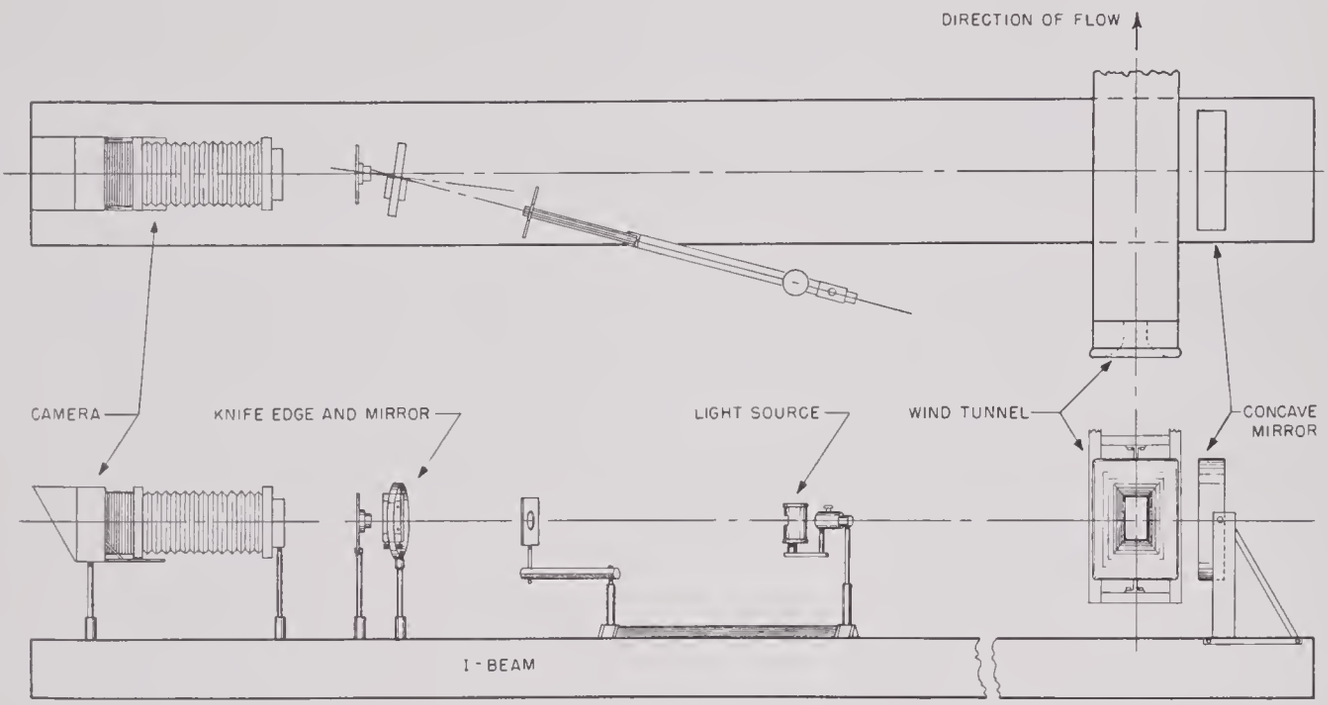


FIGURE 2. Striation system with high-speed wind tunnel.

The double passage of light through the striae is advantageous in that it doubles the sensitivity of the system for small deflections of the light beam. In cases where this no longer holds, the incident and reflected beams no longer traverse the striae at the same portion,

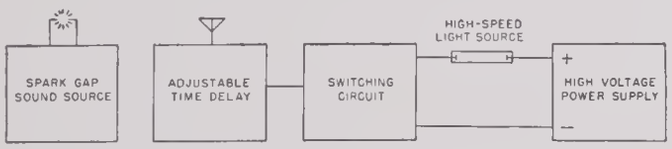


FIGURE 3. Block diagram of electronic equipment for high-speed photography.

and a double image results. Successful operation of the coincidence system depends, among other things, on locating the subject plane as closely as possible to the spherical mirror M' and avoiding the phenomenon of split images discussed above.

spark-gap sound source located in the wind tunnel, in which the desired conditions of flow have been set up. The adjustable *time delay circuit* is synchronized with the start of the sound wave by means of the electromagnetic radiation from the spark-gap wires. The delayed pulse output of the delay circuit triggers the *switching circuit* which in turn flashes the *high-speed light source*, supplied by the high-voltage power supply. This source furnishes the light to the optical striation system and a picture of the sound wave is obtained after a known time interval has elapsed from the instant the spark was fired.

The spark-gap sound source consists essentially of a power supply unit, a Ford spark coil and the spark gap itself. Its electrodes are made of pure tungsten wire 0.030 in. in diameter, ground to pin-point ends. The intensity of the sound wave from the spark depends

on the voltage and capacitance across the gap, the shape of the electrodes and their separation. The quantitative relationship existing between these variables is as yet unknown. In the region near the gap the sound waves have abnormally high velocities of propagation. It has been found, for example, that the sound wave from a 5-mm gap (about the largest used) reaches normal velocity of propagation 2.3 cm from the gap.

The time delay circuit is designed to permit time delay settings ranging from a few microseconds to about 300 μsec. The core of the device

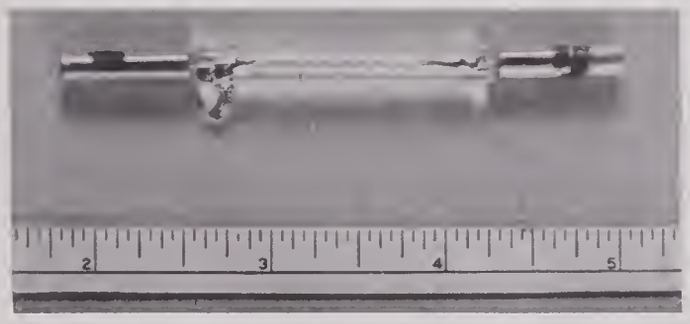


FIGURE 4. Flash light source (GE Type H-6).

is an arrangement of two tubes in a trigger circuit which has come to be known as the "flip-flop" circuit of the positive grid type. It is closely related to the Eccles-Jordan trigger circuit except that it has only one position of stable equilibrium. With suitable precautions and careful design it is possible to obtain a linear relationship between the settings of a capacitor and the time delay between the input and output pulses of the device. The input pulse is the pulse of electromagnetic radiation from the spark gap. The output pulse is a surge generated by the trigger circuit returning to its stable equilibrium.

This output pulse is passed on to the switching circuit which consists essentially of an amplifier and a thyratron switch. When the thyratron grid is driven positive, the high voltage on the plate causes it to fire and the high-voltage power supply discharges through the light source, producing a brilliant flash of less than one-microsecond duration. Thyratrons of the FG-17 and 4C35 types have been used with success.

The General Electric Type H-6 lamp is suitable as either a continuous or a flash light source for schlieren systems. This lamp is a quartz capillary tube 3¼ in. in length and ¼ in. in outside diameter with mercury pool electrodes (see Figure 4). For operation as a flash source of light a voltage of 5 to 6 kv is necessary. The light intensity is comparable with the intensity of a carbon arc.

A series of photographs of a sound wave in still air taken for a number of time delays increasing in succession is shown in Figure 5.[b] For comparison, a gauge block 5.00 cm wide has been included in the photographs. Such a set of photographs can be used to calibrate the electronic equipment directly in terms of increment of capacitance (in the trigger circuit) per microsecond difference in time delay. The capacitance values C corresponding to the different time delays t were determined by a precision capacitance bridge. The corresponding radii of the sound waves were carefully measured by comparison with the image of the gauge block, using a traveling microscope. From a measurement of the temperature the sonic velocity can be computed[3] and hence the overall calibration can be determined. Five determinations of the constant of proportionality between C and t over a period of several months yielded the value 29.25 μsec per 0.001 μf $\pm$ 0.4 per cent. From other observations it was also ascertained that, for all practical purposes, the time vs capacitance line passes through the origin.

Figure 6 shows the set of experimental points corresponding to the photographs shown in Figure 5. The slope of the curve at any point is the sonic velocity at that distance from the spark gap. For example, at a distance of 0.55 cm from the source the wave in Figure 5 is traveling with a velocity nearly 50 per cent larger than the normal value.

17.3.3 Studies in a High-Speed Wind Tunnel: Results and Conclusions

The conclusions formulated in this section are based on the results obtained up to Septem-

[b] The fact that condensations and rarefactions show dark and light on one half of the pictures and reversed on the other can easily be explained from the optical properties of the schlieren system used.

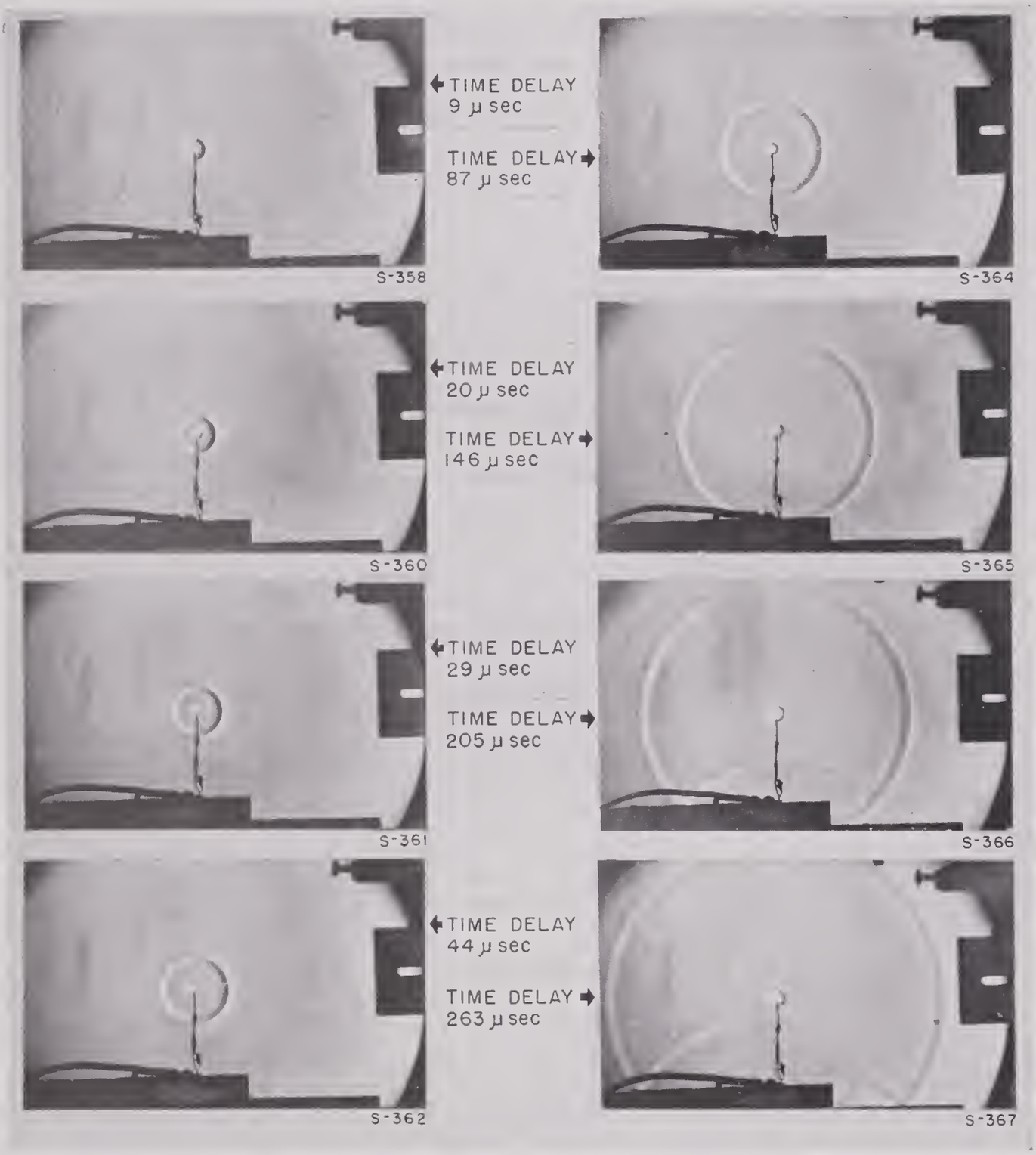


FIGURE 5. Striation photographs of sound waves in still air.

ber 1945 and are to be regarded as tentative. Most conclusions are based on a qualitative study of a considerable amount of striation photographs of sound waves in moving air. They are indicative of further problems awaiting solution.

In Figure 7 a striation photograph[c] of a sound wave is shown propagating in an air stream of extreme turbulence, which was caused by the particular spark-gap arrangement used. It is evident that the wave front has been broken up by turbulence, but it can still be located without difficulty.

Figure 8 shows a sample of photographs taken with a sound source consisting of five gaps connected electrically in series and mounted in a bakelite strip. The bakelite strip was mounted flush in the floor of the tunnel.

From these and other photographs the following conclusions can be drawn.

1. Fine-scale turbulence, such as is observed in the striation photographs, does not seriously interfere with the propagation of sound waves from spark gaps. As the turbulence increases

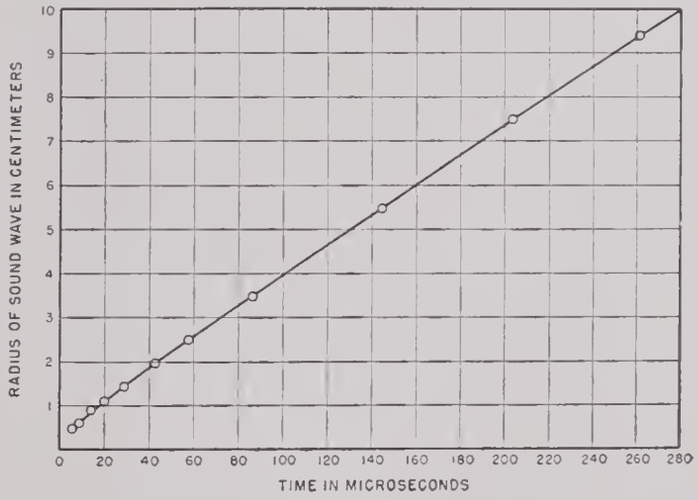


FIGURE 6. Radius of sound waves as a function of time.

the definition of the wave fronts becomes poorer.

2. At subsonic velocities, sound is propagated both upstream and downstream along a flat surface over relatively short distances only.

3. The density gradient produced by the sound wave is, in general, higher at a point upstream than at the corresponding point downstream. The ratio of the downstream density gradient to the upstream density gradient is $(1 - M)^2/(1 + M)^2$, for the subsonic case. The upstream wavelength[d] is

FIGURE 7. Striation photograph of sound waves in a turbulent stream.

shorter than the wavelength downstream. The ratio of the two is $(1 - M)/(1 + M)$.

The problem of the propagation of acoustic shock waves in a boundary layer where a velocity gradient exists has been approached theoretically. The following conclusions can be drawn.

1. The sound upstream from the sound source is refracted away from the surface by the boundary layer. This refraction results in a significant decrease in the sound level at the surface.

2. The sound downstream of the sound source is refracted toward the surface by the boundary layer. This refraction results in total internal reflection for a sizeable portion of the sound waves. The reflected rays return to the surface and are reflected back into the boundary layer where they again undergo total internal reflections. This "trapping" of the rays will result in interference phenomena for continuous waves.

It should be noted that, although important

[c] Note that in all photographs the air stream is flowing from right to left.

[d] The wavelength is taken to be the distance from the leading edge of the compression wave to the trailing edge of the rarefaction wave. It is of the order of 1 cm here.

in their own right, some of the above conclusions are not directly applicable to a practical STASI installation using sound waves of a wavelength many times as long as the length of waves investigated in the wind tunnel.

The question is not yet settled as to whether the turbulence around the source and microphones in a practical installation on an aircraft behaves in a manner acoustically similar to that of the fine-scale turbulence present in the tunnel. As to the propagation of long-wave sound along an airfoil in high turbulence, it can only be conjectured that if propagation takes place it comes about not by direct propagation in the boundary layer, but by spreading into the layer of Huyghens wavelets from the wave external to the layer.

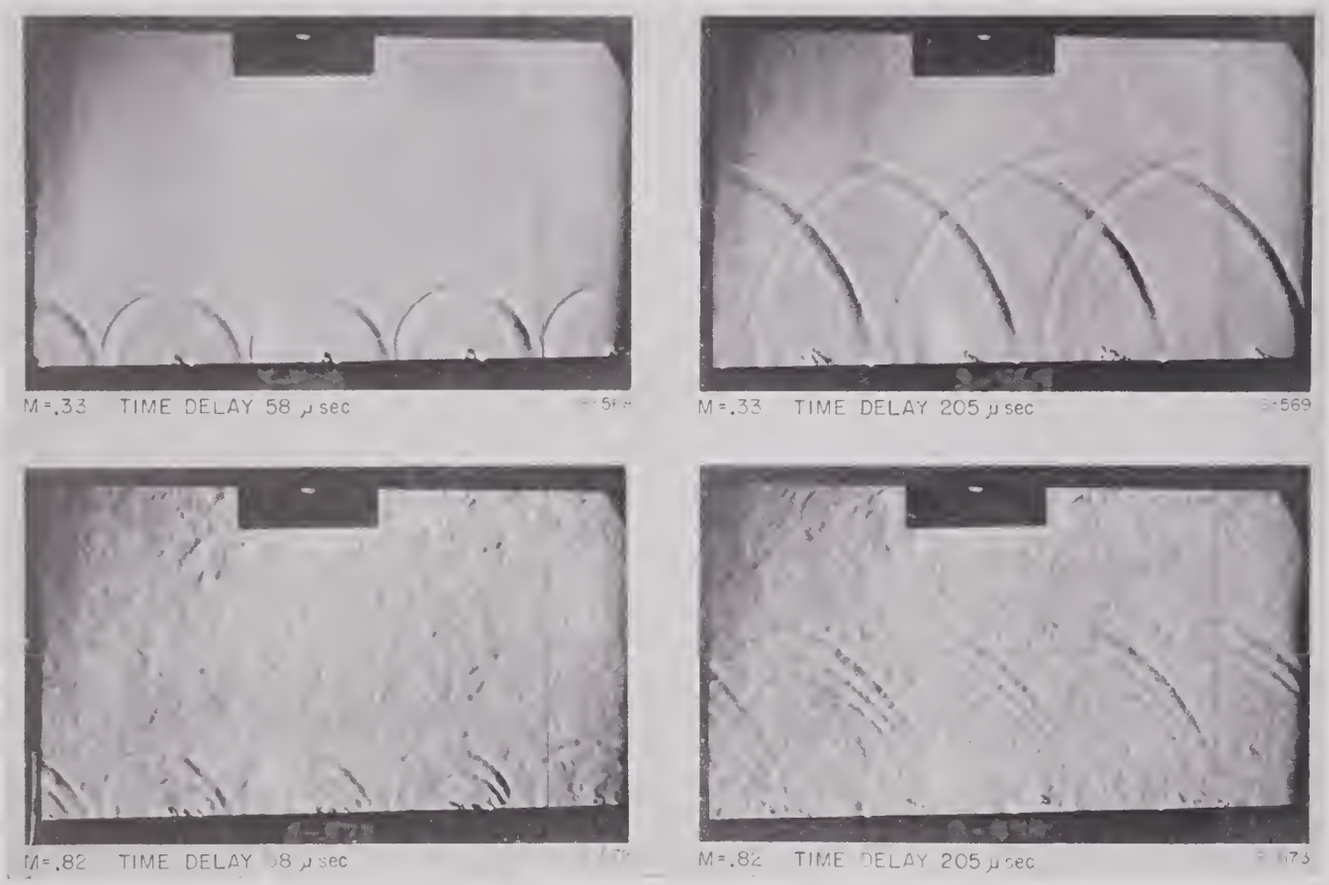


FIGURE 8. Striation photographs of sound waves from a multiple spark gap in moving air.

17.4 DESIGN AND TESTS OF A STASI FOR USE ON A NAVY BLIMP

A STASI was designed based on the principles discussed in Section 17.2 and successfully flight-tested.[5] Its feasibility at low airspeeds had been demonstrated by the wind tunnel tests discussed in Section 17.3.1.

From Table 1 in that section it is evident that, for a STASI using a continuous wave signal and a symmetrical array, a measurement of the outputs of the two microphones relative to the input to the sound source is required if the indications are to be independent of the temperature. In addition, frequency and microphone separation must be known.

The general operation of the experimental STASI will now be described. (A detailed discussion of the circuits can be found elsewhere.)[5a] Since the flight tests were conducted for the sole purpose of obtaining experimental data no attempt was made to minimize size and weight of the electric equipment. Proposed modifications desirable for the construction of a preproduction model are given at the end of this section.

A block diagram of the laboratory model of a STASI is given in Figure 9. An oscillator,

operating at a frequency of 4,000 c, provides the electric power to drive the sound source and to provide the reference voltage for the measurements of phase angle. The sound source and the two microphones are in an external sound head mounted in a suitable position on the airship. From each of the two external microphones which receive the airborne sound the signal is passed through an amplifier with automatic volume control to a phase discriminator unit.

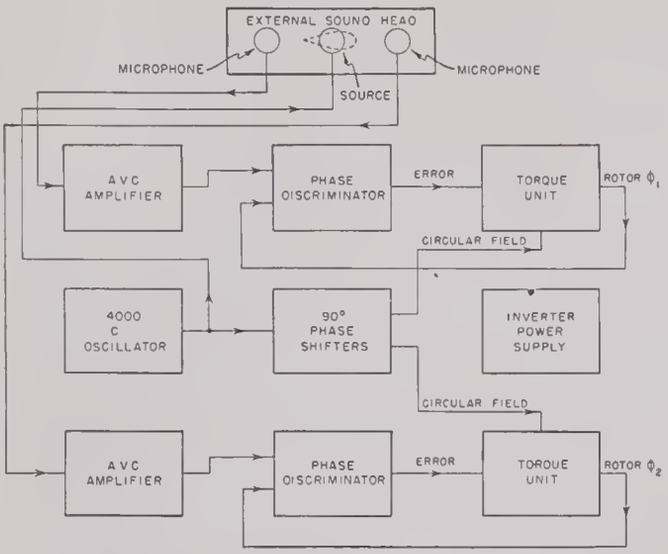


FIGURE 9. Block diagram of sonic true airspeed indicator.

The phase angle indicator makes use of a Bendix torque unit which contains a reversible motor geared to a precision Autosyn with 90-degree field windings. The output signal of the oscillator is used to feed the two stator windings of the Autosyn with voltages shifted by 90 degrees in phase. A rotating circular magnetic field is formed by the two stator coils and the phase of the voltage induced in the rotor winding depends upon the orientation of the rotor in the circular field. This voltage is fed into the phase discriminator circuit and compared with the microphone output voltage. If the phase angle between these two signals is not a fixed value (90 degrees), an error voltage is generated by the phase detector. This

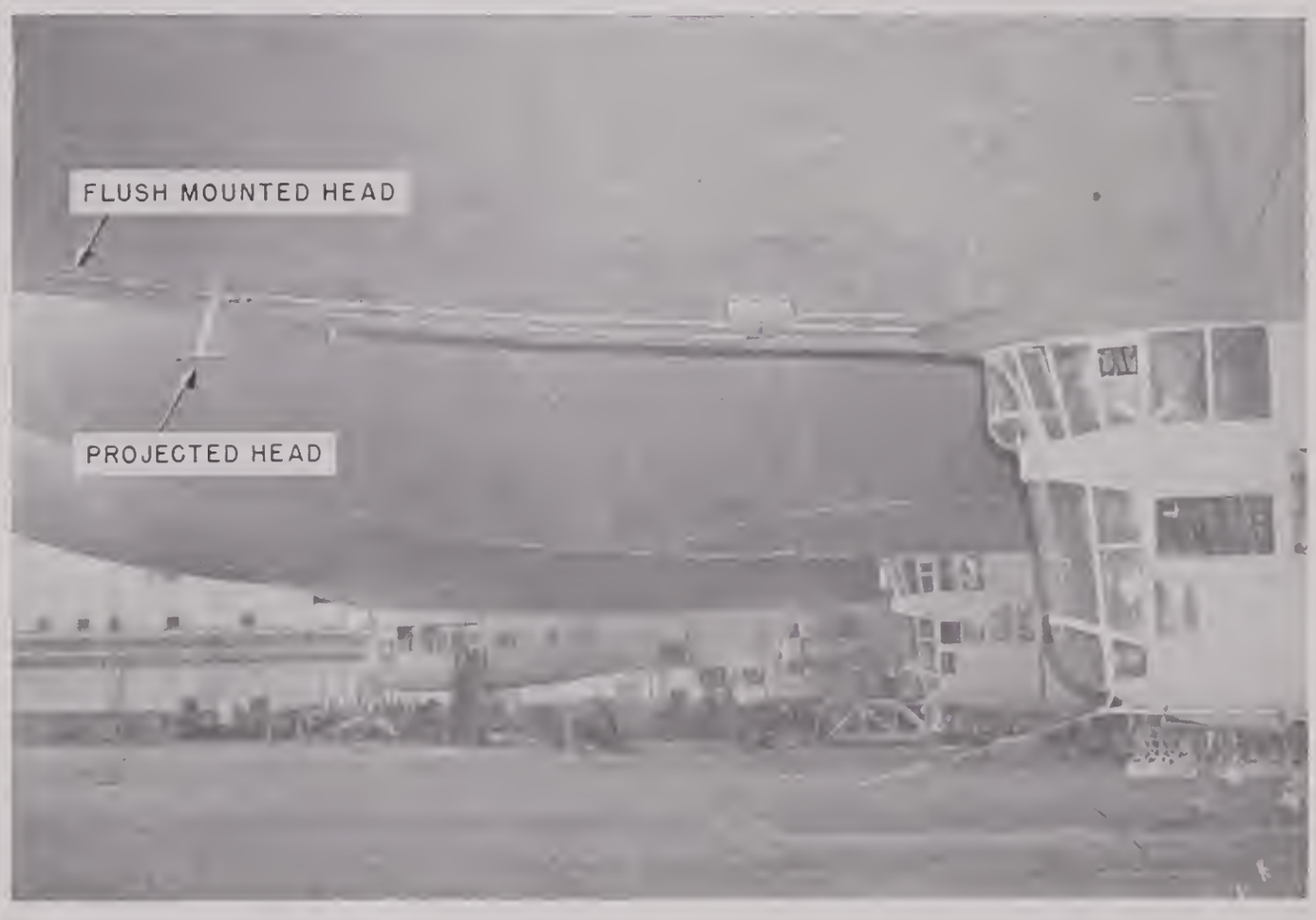


FIGURE 10. Sonic arrays mounted on Navy K airship. (U. S. Navy photo.)

voltage actuates the motor in the torque unit which turns the rotor of the Autosyn until zero-error voltage obtains. In this manner, a pointer mounted on the rotor shaft gives a direct and linear measure of the phase of the microphone output signal with respect to the reference voltage.

The experimental apparatus consists of two

Each of the arrays contains a dynamic earphone (Permoflux PDR-10) as the sound source, flanked by two modified Western Electric Type MC-253 magnetic microphones.

Figure 12 shows the projected head in detail. A typical frequency-response characteristic of the microphones is given in Figure 13. The microphones are operated in an "off-resonance"

FIGURE 11. Detailed view of sonic arrays mounted on Navy K airship. (U. S. Navy photo.)

of these phase-measuring channels, one for each microphone. The two phase angles ϕ_1 and ϕ_2 were read separately and the airspeed calculated from these readings.

Figures 10 and 11 show how two experimental sonic arrays (heads) were mounted on a Navy K-24 airship. The projected head extends 27 in. below the envelope on a streamlined strut, while the flush-mounted head can be seen cemented directly to the envelope with a fabric patch.

condition which facilitates the matching of the two units.

Figure 14 gives an estimate of the noise from aerodynamic and other causes picked up by the microphone for certain flight conditions. The cross on the graph corresponds to the signal. These data show that filters in the microphone channels are superfluous under these circumstances.

Figure 15 shows the electronic and indicating equipment mounted near the navigator's

table. Details of the phase angle dials and Bendix torque unit are apparent from the photographs of Figure 16.

The equipment described above was calibrated and tested by flying along a straight-line ground course in both directions under

FIGURE 12. Projected sonic head.

controlled flight conditions. During each test run, record was made of the elapsed time, drift angle, airspeed measured by pitot-static means, airspeed measured by two Bureau of Standards heads (one trailing and one mounted near the

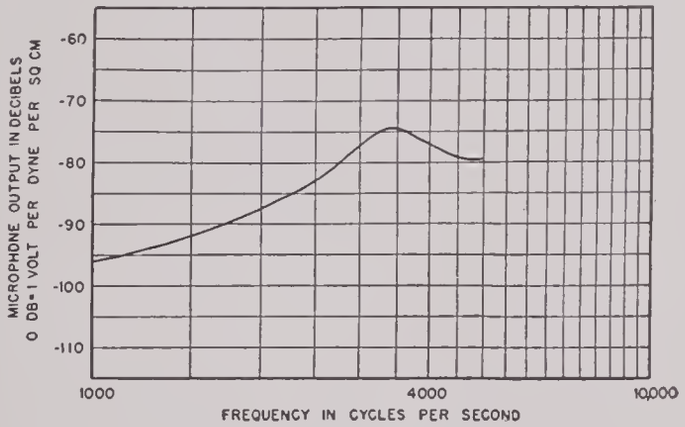


FIGURE 13. Frequency response of microphone.

sonic head), and the two phase angles of the sonic indicator. The time required to fly the course of 2.615 nautical miles was determined by means of aerial photographs of the ground course and a clock. The oscillator frequency was checked periodically in flight and the phase angles for zero airspeed were measured in the hangar prior to the flights.

From the drift angle, flight time, and course length the average "clocked" airspeed can be computed. The controlling error in the computed clocked airspeed is the error resulting from nonideal flight conditions of fluctuations in wind and variation in heading angle. The probable error due to these causes is estimated at 2 per cent. The readings of the pitot-static and standard heads were taken for check purposes only.

Figure 17 shows the average airspeed computed from the indications of the STASI plotted against average clocked airspeed. The departure of the experimental points from the straight line of optimum fit is less than 2 per cent. The horizontal line through each point is an indication of the standard deviation of the clocked airspeed for a fixed engine power, and the vertical line indicates the standard deviation of the sonic airspeed for a number of runs.

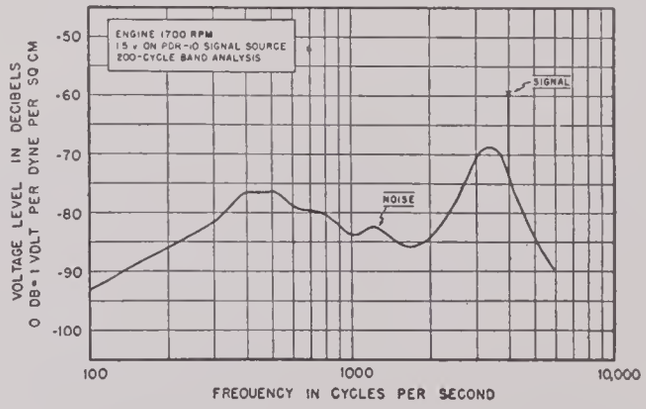


FIGURE 14. Noise analysis of microphone output.

The dotted line would be obtained if the STASI read clocked airspeed directly. The deviation in slope (about 5 per cent) between the two lines is due to the installation error. The reason for the intercept at zero airspeed is not fully understood as yet.

The foregoing results were obtained with the *projected head*. The array mounted flush with the envelope of the airship proved unsatisfactory.

By further development the bulk and weight of the equipment described above can be materially reduced. The sonic arrays can be simplified if the temperature independence is

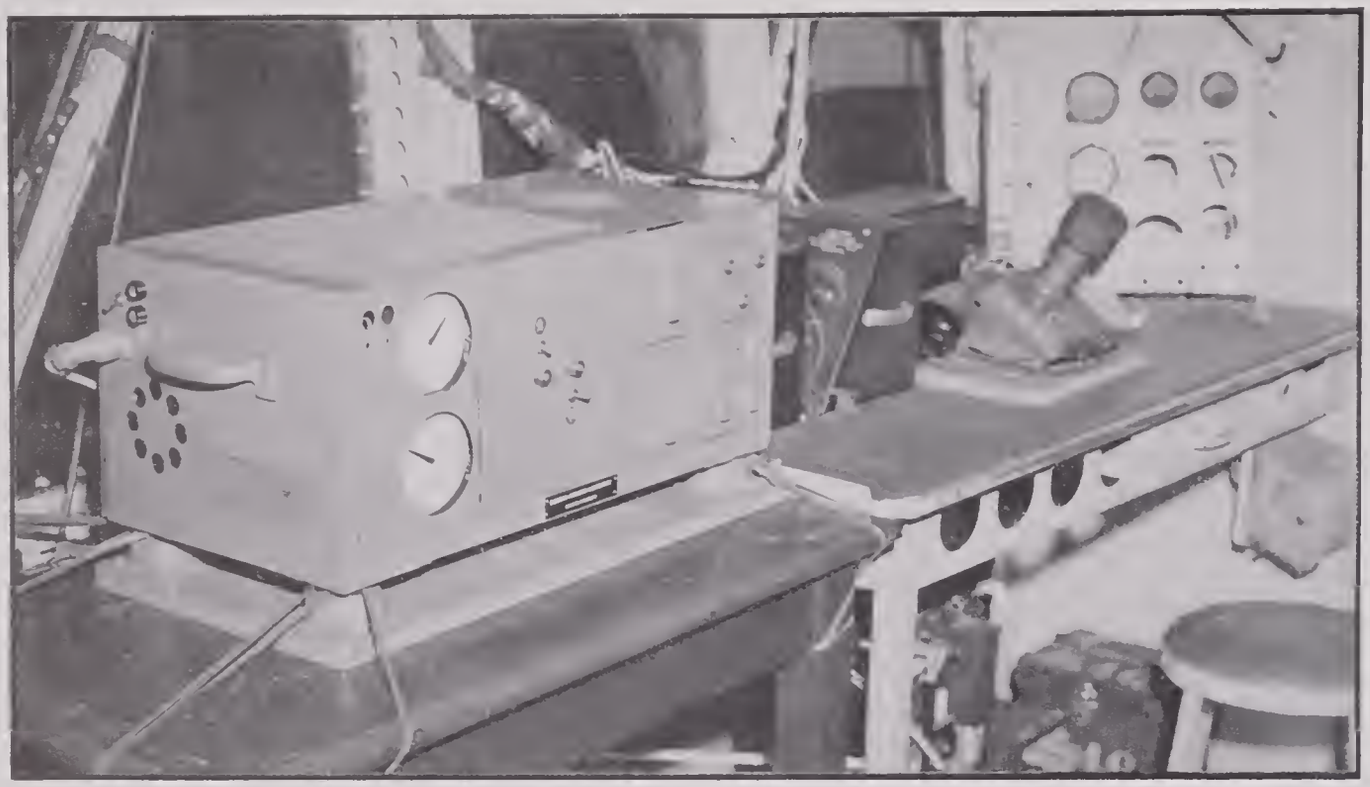


FIGURE 15. Experimental STASI indicator installation on Navy blimp. (U. S. Navy photo.)

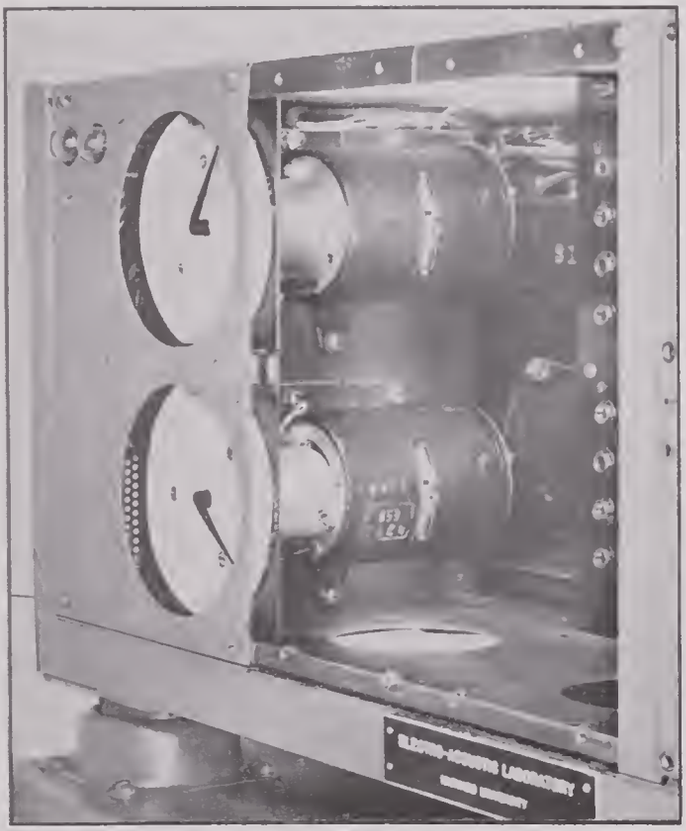


FIGURE 16. Phase angle dials with torque units.

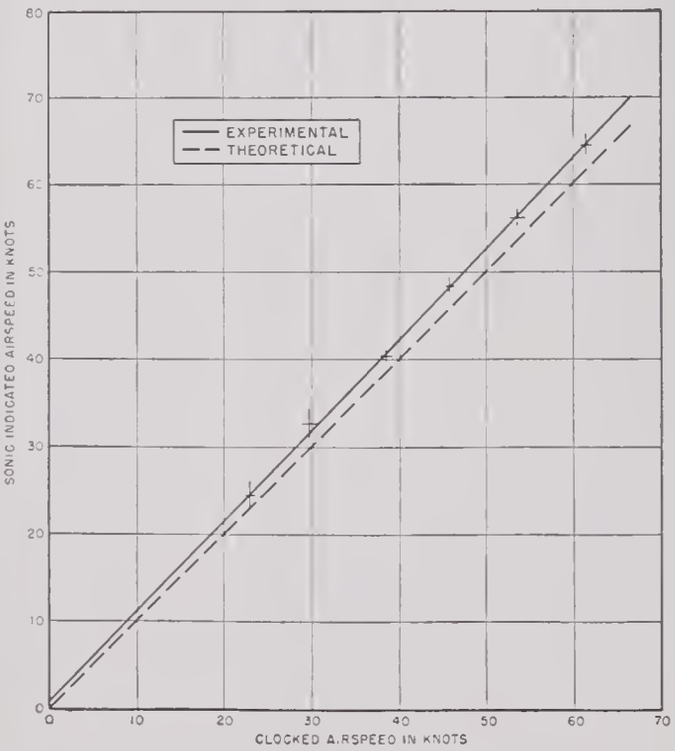


FIGURE 17. Plot of average indicated airspeed (projected head) vs average clocked airspeed.

sacrificed. The two phase angle indications can be combined into a single indication of a suitable instrument, calibrated directly in miles per hour. This can be done by electrical or mechanical means. A mechanical solution is given elsewhere.[5b]

17.5 DETERMINATION OF THE VELOCITY DISTRIBUTION IN WIND TUNNELS BY A SONIC METHOD

Successive photographs of an acoustic pulse wave with various known time delays between successive positions of the wave front can be used to determine velocities and temperature in wind tunnels. Since the spark gaps generating the sound can be accommodated in many cases with a minimum of stream disturbance, the sonic method may constitute a useful tool in wind tunnel research.

In the case of parallel (one-dimensional) flow a simple procedure can be followed. The spark gap is mounted in the tunnel downstream of the region of interest. A series of successive snapshots of the sound waves is then taken. The slope of the distance vs time plot of the wave fronts along the line of flow is equal to the velocity of propagation relative to the tunnel, namely, the difference between the local sonic velocity and the stream velocity. From this relation and the Bernoulli equation the two unknowns, namely, the stream velocity and the sonic velocity (and hence the free stream temperature), can be determined. A temperature measurement of the undisturbed air feeding the tunnel is also required. The estimated accuracy in computing Mach numbers from these data is ± 1 per cent.

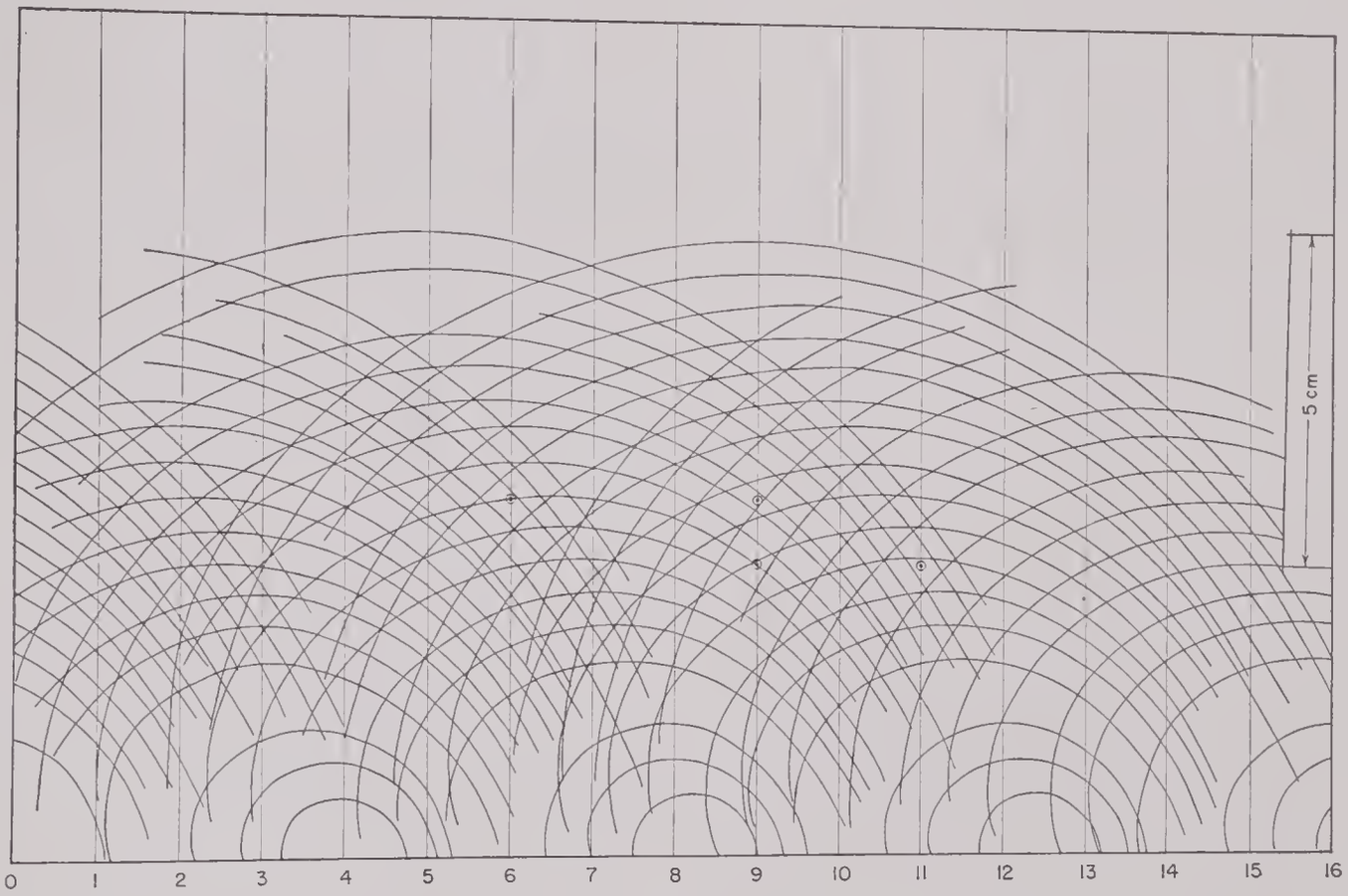


FIGURE 18. Tracing of photographed wave fronts.

Two other methods can be used which are applicable to two-dimensional flow. They yield the two components of the stream velocity and the sonic velocity (and hence the temperature) as a function of position in the stream.

It is assumed that the motion of the acoustic disturbance is the same as though at each moment each point on it were moving with

a velocity compounded of (1) the wave velocity c at the point in question and (2) the stream velocity at that point.[1] With these assumptions it can be shown that the following equation (wave-front equation) holds:

$$-\frac{\partial y}{\partial t}+c\sqrt{1+\left(\frac{\partial y}{\partial x}\right)^2}=-v+u\frac{\partial y}{\partial x}, \quad (1)$$

where x and y are the two coordinates, u and v the two components of the stream velocity in the direction of the two coordinate axes, c the local sonic velocity, and t the time.

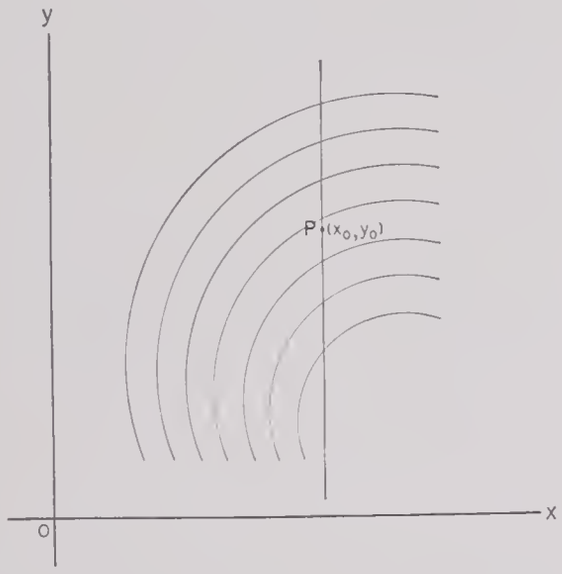


FIGURE 19. Details of tracings of photographed wave fronts.

Let us assume that many photographs of sound waves generated by a multiple spark gap (such as shown in Figure 8) are available, taken at successive known time intervals. If all pictures are superimposed and traced, a grid such as shown in Figure 18 will result. Define the coordinate axes and draw a line through the point of interest $P(x_0,y_0)$ parallel to the y axes. (See Figure 19.) Along this line the distances above the x axes, at which successive wave fronts originating at the same source intersect the line, can be measured. Since the time increments are known, a curve y vs t for x_0 = constant can be drawn and hence $\partial y/\partial t$ at x_0 determined. The slopes, $\partial y/\partial x$, of the wave fronts where they intersect the line $x = x_0$ can also be measured, and by interpolation, $\partial y/\partial x$ at P can be calculated.

This procedure is repeated for two other families of wave fronts emanating from two other spark gaps. Substituting the three pairs of values of $\partial y/\partial x$ and $\partial y/\partial t$ into equation (1) three simultaneous equations, linear in u, v, and c, are obtained which can be solved easily for the three unknowns at $P(x_0,y_0)$.

In the second method based on the wave front equation (1) only two families of waves are evaluated as described above. As the third equation the Bernoulli equation is used. A knowledge of the temperature of the undisturbed air before entering the tunnel is required.

The estimated probable error in the determination of the principal stream velocity is about 15 per cent. It is believed that this figure can be improved upon by improvement of the experimental procedures involved. The merit of both methods lies in the fact that the spark gaps can be mounted flush with the tunnel bottom and hence the disturbance to the stream can be made negligible.

Chapter 18

AUDITORY SIGNALS FOR INSTRUMENT FLYING: "FLYBAR"

THE LARGE NUMBER of operations required of the pilot of a modern airplane and the multitudinous instruments which he must follow tax his abilities to the limit. His eyes especially are taxed, since he must keep track of the large number of instruments on the panel and at the same time see obstacles, other airplanes, and the ground at critical times. In the larger airplanes both pilot and co-pilot are sometimes kept busy, especially when flying under instrument conditions.

The continuously increasing speed of modern planes and the advent of new devices such as airborne radar will, if anything, increase this critical load on the eyes. Thus it would be an advantage if some of the flight indications could be furnished to the pilot by way of his ears.

As early as 1936 it was demonstrated[1] that a pilot could fly an airplane with his eyes blindfolded when two instrument indications were given by means of an auditory signal in his earphones. The pilot flew with rudder and elevators only, allowing the inherent stability of the Fairchild 24 to take care of lateral control. The signals employed were (1) a turn indication consisting of increase of the signal intensity to one ear and decrease of intensity to the other, and (2) an airspeed indication consisting of a change of pitch of the signal. Although the plane described a wide climbing spiral rather than a straight path, the pilot was able to maintain satisfactory control and to recover from spins, thus showing the feasibility of flying by auditory reference, *flybar*.

The purposes of the present study[2] were to determine (1) what types of auditory signals could be followed with greatest ease, (2) with what accuracy such signals could be followed, and (3) how many simultaneous auditory signals could be followed successfully. It was, therefore, necessary to design a variety of different auditory signals and to test their effectiveness on a group of men who were performing a task similar to that of "flying blind" in an airplane.

The results indicate that if the signals are properly designed as many as four auditory indications can be followed without interfering with radio and interphone communication when occasion demands. It is thought that such signals may be of value not only in military aviation but also in peacetime flying.

18.1 DEVELOPMENT OF AUDITORY SIGNALS

18.1.1 Tone Signals

The aim of this part of the investigation was to develop three aural indications which could be followed simultaneously. Indications for turn, bank, and airspeed were tried out on two synthetic devices. One of these was the airplane pursuit meter, a device in which the subject tried with controls similar to those in an airplane to compensate for deviations introduced in an unpredictable fashion by a mechanical device. Automatically computed scores indicated the success with which the signals were followed. The second device used was the familiar Link Trainer, fitted with auditory signals to give turn, bank, and airspeed indications. The records obtained by use of the auditory signals were compared with records made by the same individuals using visual indications.

Ten men between the ages of nineteen and thirty-six were used as subjects. Since none of these men had had previous flying training, the course of learning could be followed for both the visual and auditory types of indication. The task, in the case of the Link Trainer, was to fly a straight course by means of turn, bank, and airspeed indications alone (without any compass or gyro direction indicator) and with the rough-air attachment turned on. The task set by means of the airplane pursuit meter was a somewhat similar one. Several trained pilots also tried the best signals on the Link Trainer.

A number of different types of auditory signals were experimented with, the first being

combinations of buzzes, pitches, "chopped" signals, and other conventional tone signals. In general, it was found that combinations formed by adding separate tone signals together were too complex to be followed successfully. Some individuals had difficulty in analyzing each individual signal in the complete auditory pattern when two or more signals were used. There was a general tendency when using these complex signal combinations for the subject to follow one indication with such attention that the other two escaped him and "went out of control."

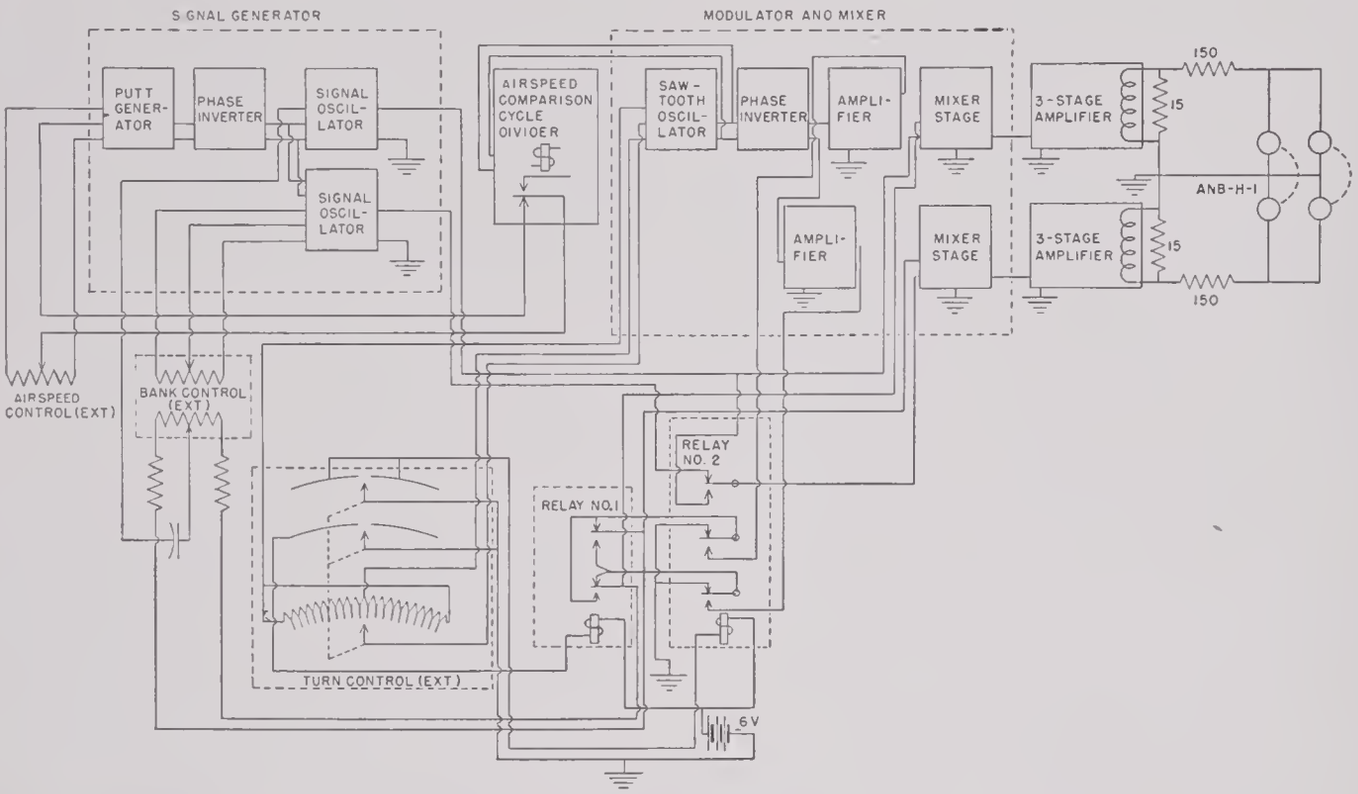


FIGURE 1. Relay connections and block diagram for the successful three-in-one auditory indication.

FIGURE 2. The Link Trainer with the signal equipment mounted for test purposes.

Two auditory signal combinations were developed, however, which were successful enough to be of possible practical utility. Both "sounded like the behavior of the airplane." One involved apparent motion of the tone to the right or to the left in a manner roughly corresponding to the right and left indications of a radio compass. This signal can be used with a second indication superimposed on it to indicate airspeed.

The second successful signal was a three-in-one indication. That is to say, different characteristics of the same signal were used to indicate turn, bank, and airspeed. These indications were, respectively, (1) a repetitive or sweeping type of motion of the signal from left to right, or right to left, (2) an apparent "tilt" produced by pitch variations, and (3) a "putt" sound, the rate of occurrence of which could be associated with the sound of the airplane motor. A block diagram of the device used to generate this signal is shown in Figure 1.

With this three-in-one signal the ten subjects learned to operate the Link Trainer so as to fly a respectably straight course in "rough air" after from 2 to 2½ hours of practice. The results corresponded quite well with those obtained by the same men using visual indications after 1½ to 2 hours of practice. Figure 2 shows the signal equipment mounted on the Link Trainer for test purposes while Figures 3 and 4 give sample records for one subject using both

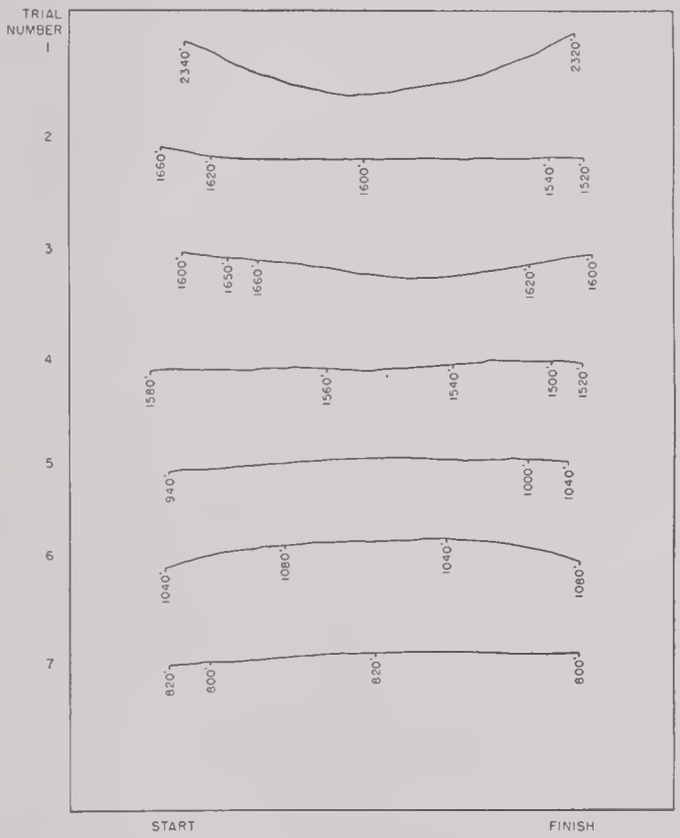


FIGURE 3. Link Trainer records for subject using visual indications.

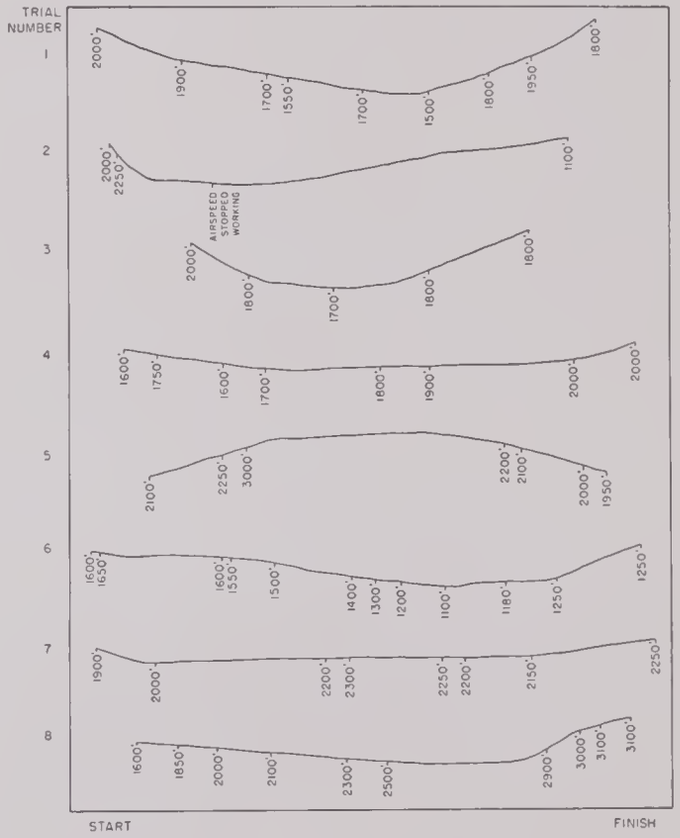


FIGURE 4. Link Trainer records for subject using auditory indications.

types of indication. As would be expected, some of the men had more difficulty than others, both in learning to manipulate the controls and in following each type of signal.

Six private pilots, some of whom had had instrument instruction, were able to operate the Link Trainer creditably by means of the auditory signals in from 1 to 1½ hours of practice. Several Navy pilots, after a short trial, were confident that it would be possible to fly with this signal combination.

The two-in-one signal did not give quite so accurate results as the three-in-one but seemed good enough to be of possible use. Either of the two signal combinations might be applied to other instruments than the ones used for test purposes. For instance, the turn signal might be of more use if controlled by the gyro direction indicator or the radio compass for keeping on course on long flights.

It was of especial interest that the simulated radio-range signal and voice communication could be heard simultaneously with the two-in-one and the three-in-one type of signal without difficulty. This was due to avoidance of a large number of tones which would cause greater interference with range and voice signals.

18.1.2 Automatically Produced Speech Signals

A device called an automatic annunciator was developed for the purpose of announcing altitude, airspeed, or other similar instrument indications directly to the pilot. This device translated the indicator readings automatically into spoken messages. It proved quite successful and

offered promise of useful application for many types of instrument indications and of warnings.

As developed for demonstration purposes, the device announced altitude in 200-ft units through the pilot's earphones. The annunciator consisted of a light, compact, multichannel sound reproducer of the magnetic tape type, on each channel of which a permanent message had been previously recorded. Each channel contained one spoken number or one unit of the message and these were selected by relays operated by a control-switch unit. This unit followed the Link Trainer altimeter by means of a self-synchronous repeater and servo units. After the appropriate message units were selected they were played in the proper sequence in order to say, for example, "two-thousand two-hundred feet."

The automatic annunciator may be coupled to instruments by servo devices in cases where the aircraft instrument does not have sufficient torque to operate the control device directly. For some applications, however, the annunciator may be operated directly (without servo unit) by means of vacuum tubes.

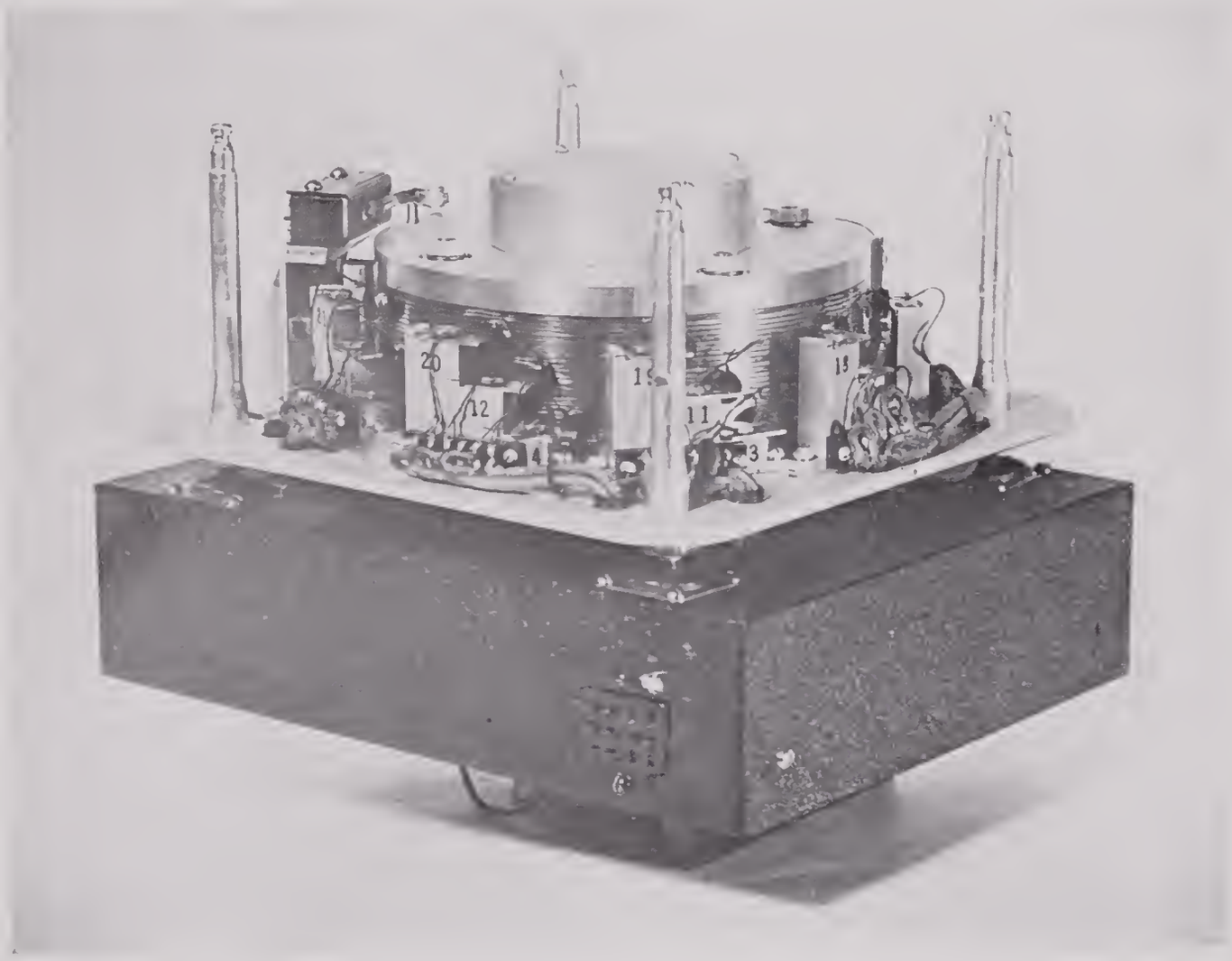


FIGURE 5. Basic assembly of multichannel sound reproducer.

The control circuit is such that when using a four-unit message such as "four-thousand four-hundred feet" the lag in the spoken indication will never be more than the length of the message plus 1 sec. In the four-unit message, this means a lag of 5 sec. When the application requires the shortest possible time interval, single-channel messages may be used to reduce

the lag to about 1 sec, or if 1-sec lag is still too long a circuit of the anticipation type can be employed.

The twenty-four channel magnetic reproducer developed for annunciator use[a] weighed approximately 15 lb. Associated selection and control equipment will weigh more or less depending on the application. Figure 5 shows the multichannel sound reproducer and Figure 6 illustrates it covered and connected to a relay

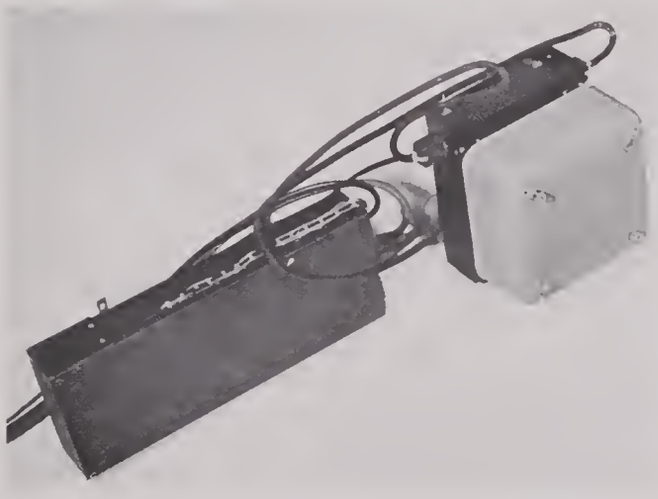


FIGURE 6. Complete annunciator—relay and sequence unit at the left, reproducer at the right.

and sequence unit developed for use with the altimeter.

It is of interest that tests with the annunciator on the Link Trainer altimeter have shown that there is very little difficulty in distinguishing between speech from the annunciator and incoming speech communication from outside (radio) channels. Communication messages may be made to override the annunciator, and, also, the annunciator has a repetitive rhythm and a speech quality which is readily identified.

18.2 FUNDAMENTAL PRINCIPLES OF SUCCESSFUL AURAL INDICATIONS

In order to be successful, auditory signals must be designed with a view to certain psychological principles related to hearing and to the pilot's reactions.

1. Pilots have certain habitual methods of thinking about the airplane. Auditory signals must be designed to fit these habitual methods of thinking.

2. Most fliers are much more accustomed to using visual indications than auditory ones. The signals should, therefore, be as simple and as self-explanatory as possible. Only under these conditions will the need for special training of pilots in use of the signals be reduced to a minimum.

3. It can be demonstrated that a person can attend independently to only one thing at a time. It was found that when several different auditory signals were added together the individual following the signals might attend for a while to each in turn, as he was supposed to do. Then, however, there was a tendency for one signal to "capture" his attention to the exclusion of the others. To be of practical value, the auditory signals must be designed so that this "capture of attention" does not occur.

4. In military aircraft, information from various sources is continuously obtained through the earphones from radio and interphone channels. Any auditory indications used must be designed to interfere as little as possible with such messages.

5. It is not sufficient to produce an auditory signal which satisfies the designer. The signals must also be tested by psychological methods on persons unfamiliar with their development. In so doing, preconceived prejudices and special abilities found among particular men will be eliminated. If this is not done, erroneous conclusions may result.

18.3 POSSIBLE APPLICATIONS

It is not anticipated that auditory signals will entirely displace visual instruments in the airplane for any of the fundamental indications. However, it has been demonstrated that as many as three auditory signals can be followed with accuracy if properly designed aural indications are used. It is thought that the spoken indications can be useful as auxiliary aids to

[a] Through the cooperation of Bell Telephone Laboratories. Appreciation is expressed to Dr. E. C. Wente of NDRC Section 17.3 and Mr. W. L. Woolf of Stevens Institute of Technology for valuable advice and interest.

furnish warnings of various types and to give altitude and airspeed information to the pilot whenever his eyes must be otherwise engaged.

A tone signal attached to the gyro direction indicator might be of considerable advantage on airplanes which do not have the automatic pilot for maintaining the proper heading on cross-country flights. Or again, such a signal might be controlled by the self-orienting loop of a radio compass for long distance "homing" on radio stations. This would allow the pilot to maintain the proper bearing on the station without the necessity of constantly watching the needle indicator of the radio compass.

It is also possible that auditory signals, if properly designed, can be of assistance in connection with some of the new blind-landing systems which are under development.

The automatic annunciator can be arranged to announce altitude or airspeed or both alternately for landings and take-offs. Such auditory messages would be of advantage in single-seater planes for landings made under conditions of low visibility. In larger airplanes the annunciator could also relieve the co-pilot of calling this information. Furthermore, the device can be arranged to call attention to sources of trouble such as "wheels not down," "flaps not down," "gas tank empty," or other such items.

In practice, of course, the slight added weight of equipment must be weighed against advantages gained. Auditory signals have a distinct advantage, however, wherever the pilot's attention must be called to a malfunctioning part or whenever his eyes must be otherwise occupied.

Chapter 19

SPECIAL DEVICES FOR USE ON SHIPBOARD

IN THIS CHAPTER a brief discussion is given of certain special equipment developed at the Electro-Acoustic Laboratory for use aboard ship. The equipment was designed primarily to facilitate and simplify the flow of information of various kinds about the ship during battle.

19.1 THE RADIO REPEAT UNIT

There are indications that present voice communication channels on shipboard may become overloaded at critical times. This is especially true of the radio channels, since they share the largest part of the burden of external communications. When messages must be repeated on the air, either because of poor intelligibility or simply because the receiving operator is not available on the particular circuit at the time, congestion results and a breakdown may occur during times of high-circuit load.

One of the simplest ways of increasing the working efficiency of a given communication channel is to eliminate all on-the-air repeats. A request was made in 1944 to the Electro-Acoustic Laboratory by the U. S. Navy Bureau of Ships to develop a simple device which would be capable of quickly repeating any radio message without interfering with the normal operation of the radio channel over which the message was received.

The *radio repeat unit*[3] [RRU] was subsequently developed at the Electro-Acoustic Laboratory, and a model was successfully tested.

The basic elements of the RRU are a recording unit consisting of an endless magnetic tape, a recording head, erasing head, and a playback head with suitable switching facilities. Figure 1 shows a simplified diagram of the recording unit. When switch S_1 is closed, any incoming signal will be recorded magnetically on the tape. Any message which might have been on it earlier is erased by the erasing head at position 3. The recorded signal, delayed by a certain time interval, is carried by the tape around the loop and played back into the earphones (S_2 closed) by the playback head at position 2. This interval of delay is determined by the length and speed of the tape. If switch S_1 is opened, all further recording and erasing ceases and the message recorded on the tape is stored for an indefinite number of future playbacks.

The tape of the required length is reeled on a magazine shown in the center of Figure 2. In operation, the tape emerges from the magazine near the center, then passes the playback, erasing and recording heads, and goes around the driving drum to be pushed back into the magazine from the outside through a guide. An alarm indicator is provided to indicate tape slack or breakage.

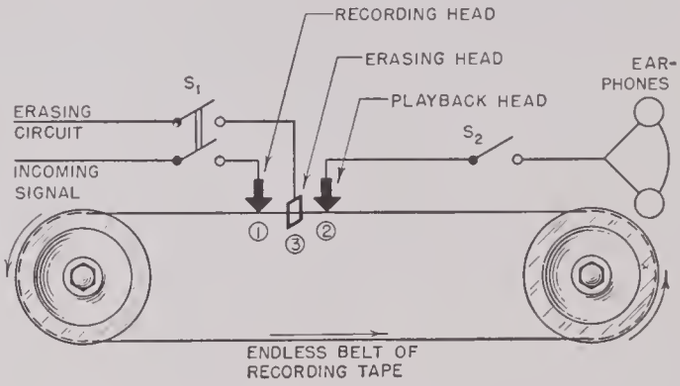


FIGURE 1. Simplified diagram illustrating the principle of the radio repeat unit.

The magazine and tape drive are of novel design, and both have performed well in life tests in the laboratory. The magazines can be easily replaced in case of defect or if a different time delay is desired.

Conventional, longitudinal, magnetic-recording technique is used with an a-c bias current in the recording head. A tape speed of about 3 ft per sec yields a satisfactory frequency response up to about 4,000 c. The tape is made of a stainless-steel alloy. A signal-to-noise ratio of at least 40 db is easily achieved.[3a]

The complete RRU has three recording loops. This makes it possible to incorporate three different delay times or storage of a message on one tape while others are being recorded on the remaining two loops. A simplified diagram of the RRU, incorporating delay times of 15, 30,

and 60 sec, is shown in Figure 3. In Figure 4 the three tape units are shown mounted side by side. A sketch of the complete unit with

Some of the useful functions of the RRU can be listed as follows.

1. Quick repeat of part or all of a message

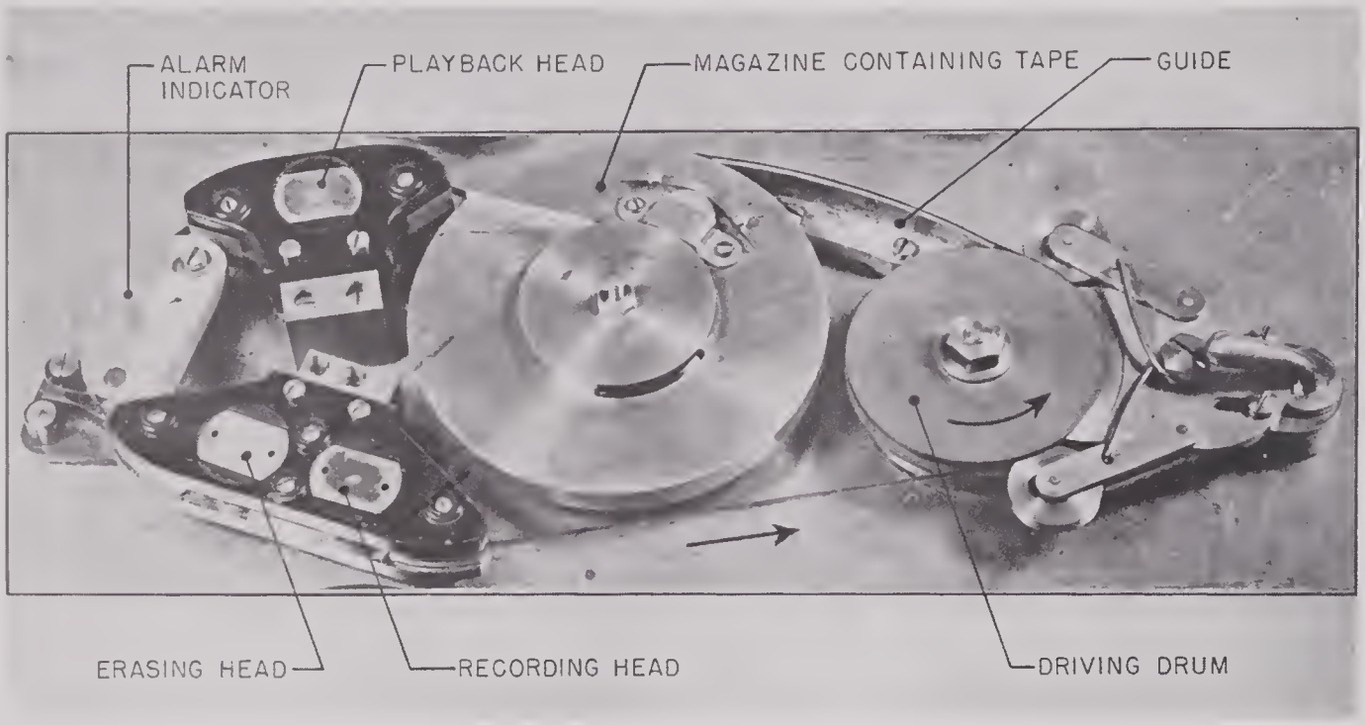


FIGURE 2. Tape unit.

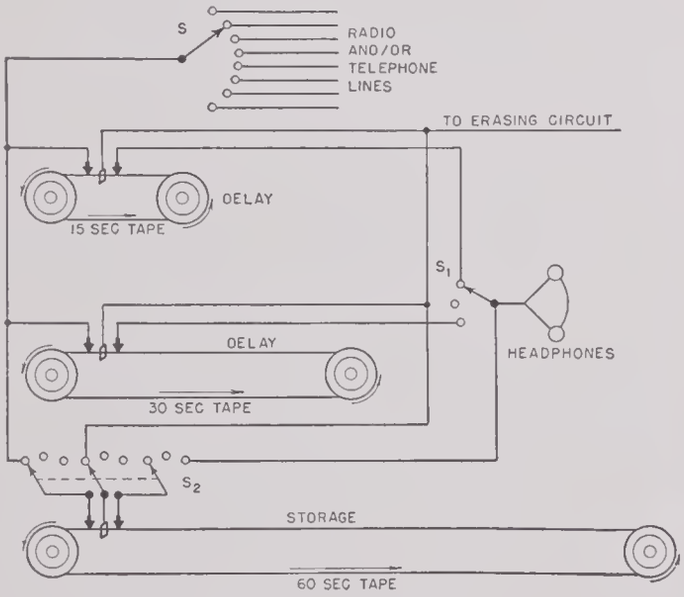


FIGURE 3. Diagram of radio repeat unit with three recording loops.

FIGURE 4. Tape units of radio repeat unit.

three remote stations is shown in Figure 5. The details of the switching arrangements can be easily varied to suit the needs of various stations and ships.

so that requests for repetition are eliminated and the circuit load is decreased correspondingly.

2. Two simultaneously arriving messages on

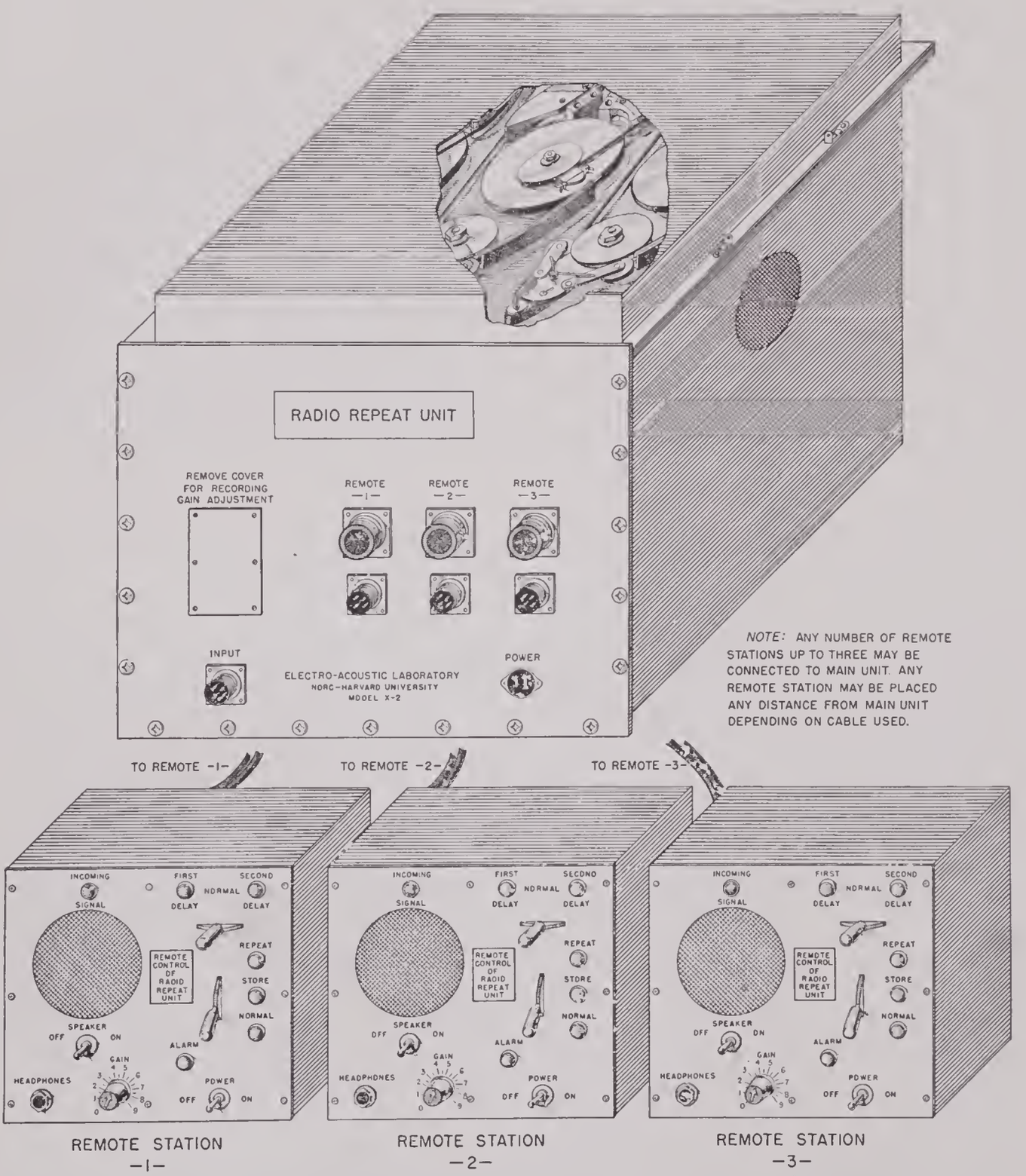


FIGURE 5. Complete radio repeat unit with three remote stations.

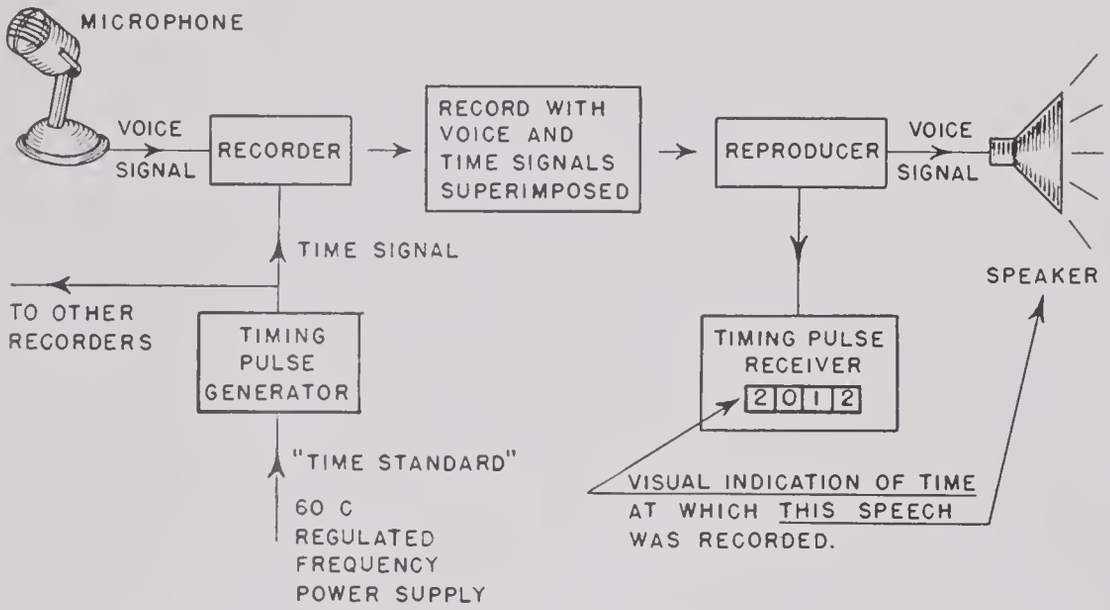


FIGURE 6. Schematic representation of the method of indicating time on voice recordings.

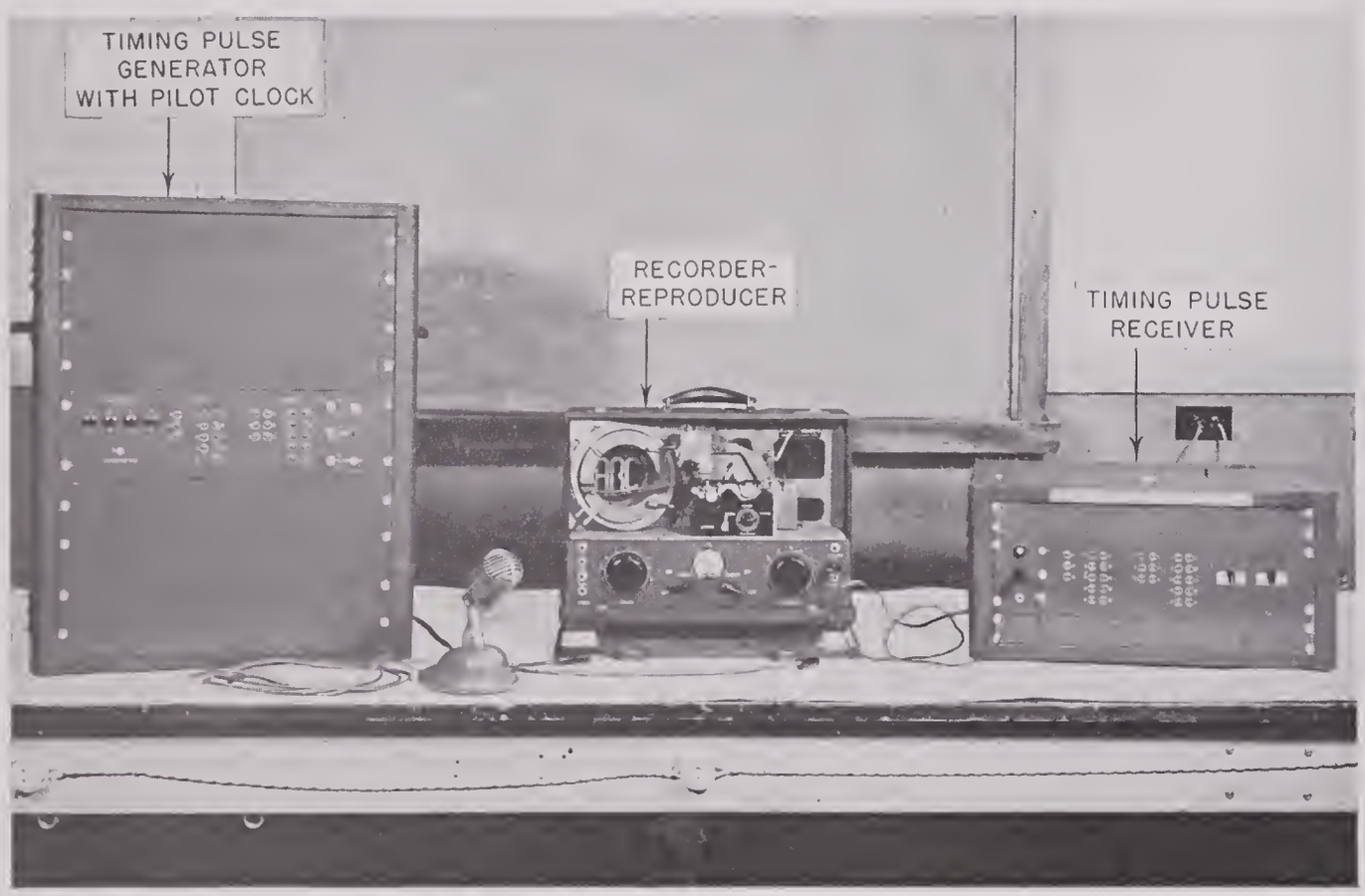


FIGURE 7. Timing pulse generator, recorder-reproducer, and timing pulse receiver (laboratory models).

two different channels for the same recipient can be separated in time by inserting a suitable time delay.

3. Temporary or permanent storage of important messages.

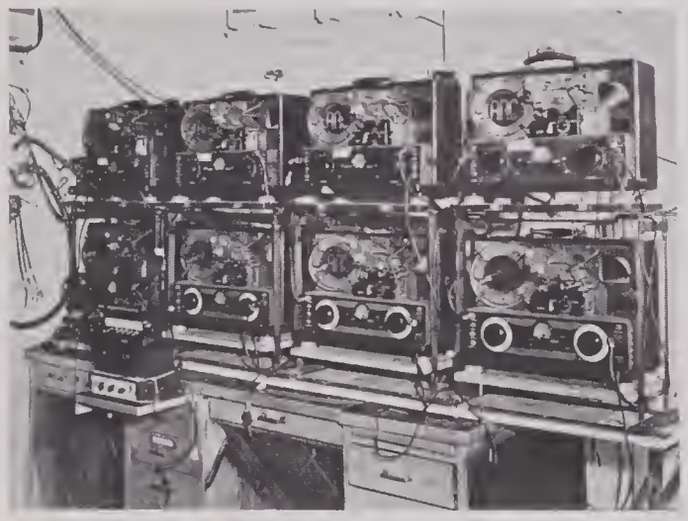


FIGURE 8. Recorders installed on shipboard.

4. Assistance to monitoring personnel in keeping complete logs, especially if more than one channel has to be supervised.

tioned above[3b] along with a detailed description of the device.

19.2 A METHOD OF TIME INDICATION ON VOICE RECORDINGS

Recordings of the intelligence transmitted by the important voice communication systems on shipboard are potentially useful in many cases. However, such recordings can be fully utilized only if an accurate indication of the time is also provided for on the record. By installing multichannel voice recording equipment on shipboard, continuous voice and time recordings can be made during action. A complete and permanent battle record is thus obtained which should prove useful in a critical evaluation of the performance of equipment and personnel.

Equipment capable of superimposing suitable time indications on voice recording without detrimental interaction and without using additional channels was designed and constructed at the Electro-Acoustic Laboratory.[1]

A schematic representation of the method of

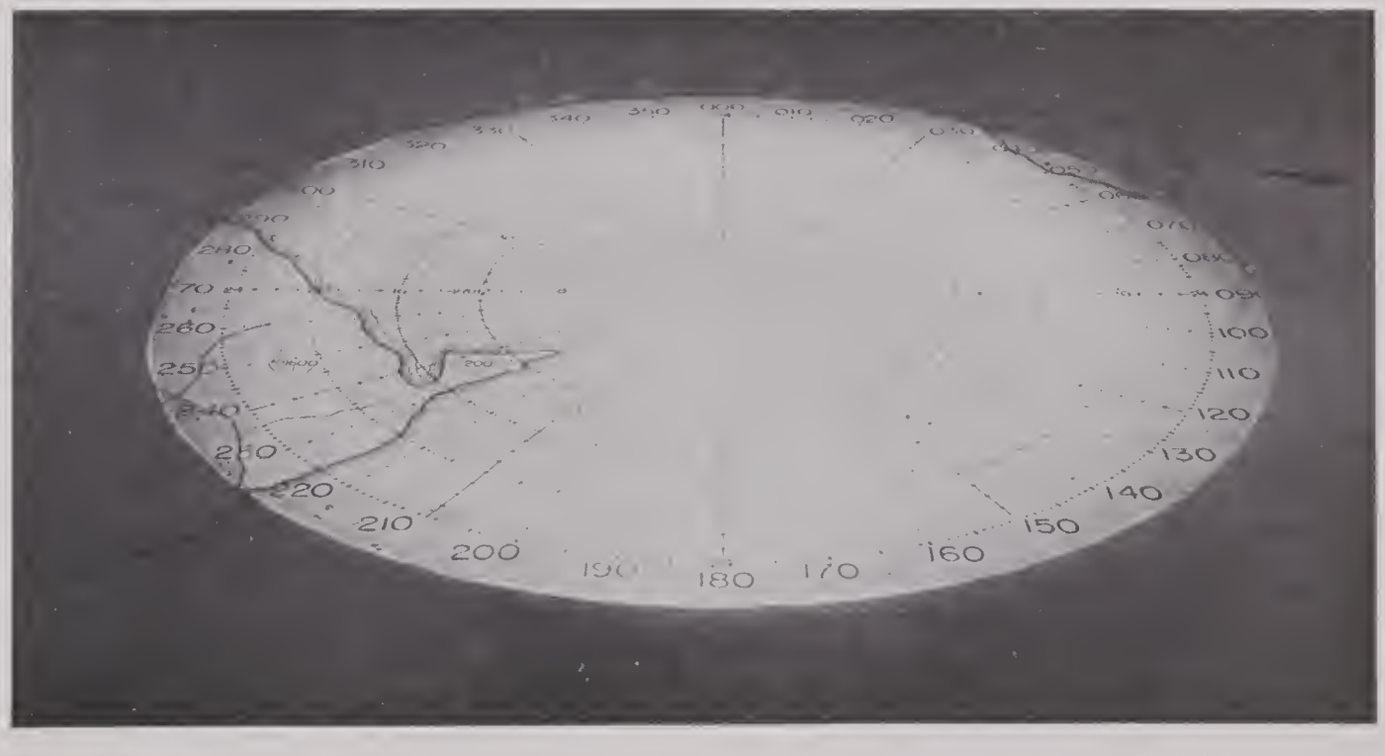


FIGURE 9. Photograph of coordinate system projected on plotting surface on DRT.

The usefulness of an RRU can be extended beyond the obvious cases given above, and such applications are discussed in the report men-

indicating time is shown in Figure 6. The power line with a controlled frequency of 60 c is used as the standard of time. Controlled by this

power line, the *timing pulse generator* [TPG] generates a suitably coded time signal which is superimposed on the voice signal and simultaneously recorded on any number of recorders. Interaction between the two signals is prevented by suitable frequency-discriminating networks. Upon playback, the time and voice signals are separated again by filters. The *timing pulse receiver* [TPR] converts the coded information of the timing pulses into a visual indication of the time at which that portion of the voice recording was made (2012, or 8:12 p.m., in Figure 6).

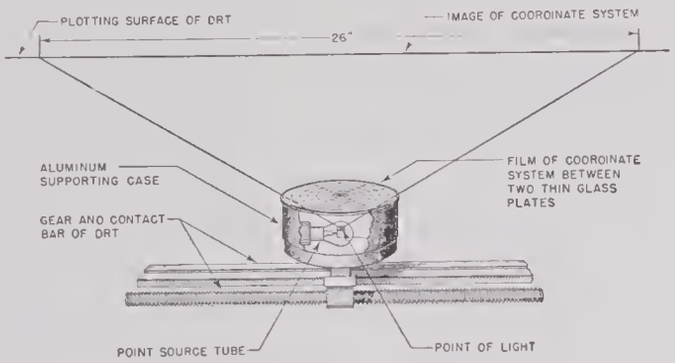


FIGURE 10. Diagram of point-source projector.

FIGURE 11. Laboratory model of point-source projector mounted on Arma class 2 DRT.

The time signal is transmitted from the TPG, not continuously but at regular intervals. The recorded indications thus form a set of "fixes," or known points on the record. Errors caused by variations in the speed of the medium are automatically rectified when the next "fix" occurs. Therefore, the errors are not cumulative, and long periods of recording and discontinuities of time in the recorded material do not affect the accuracy of the indications.

As the number of indications per unit time is increased, approaching in the limit a continuous indication, the errors accumulated between "fixes" become negligible. One-minute intervals were chosen as a practical spacing.

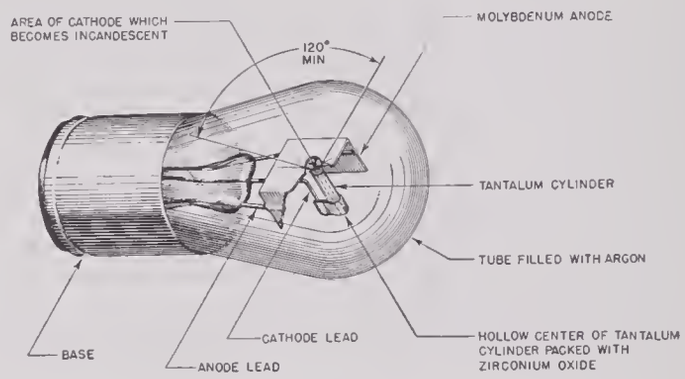


FIGURE 12. Construction of point-source tube.

Exactly once every minute, then, the time is identified on a 24-hour basis by suitably coded pulses generated in the TPG. The 1,440 minutes are identified by four groups of pulses.

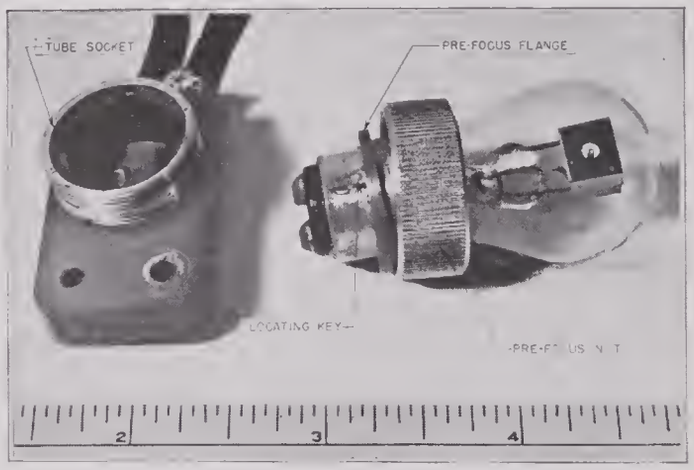


FIGURE 13. Photograph showing point-source tube.

In order to record a series of pulses with conventional equipment, a suitable "carrier" frequency was chosen which is modulated by the series of pulses comprising the time signal. By use of suitable high-pass and low-pass filter, the carrier frequency is accommodated slightly

above the upper cutoff frequency of the voice channel.

Laboratory models of the TPG and TPR are shown in Figure 7. A commercial recorder-reproducer, using an endless plastic tape as recording medium, was modified to suit the application at hand. Continuous recordings of over 1 hour duration were made in the laboratory and on shipboard. Figure 8 shows a battery of recorders installed on shipboard for an analysis of the circuit loads carried by the various voice communication channels.

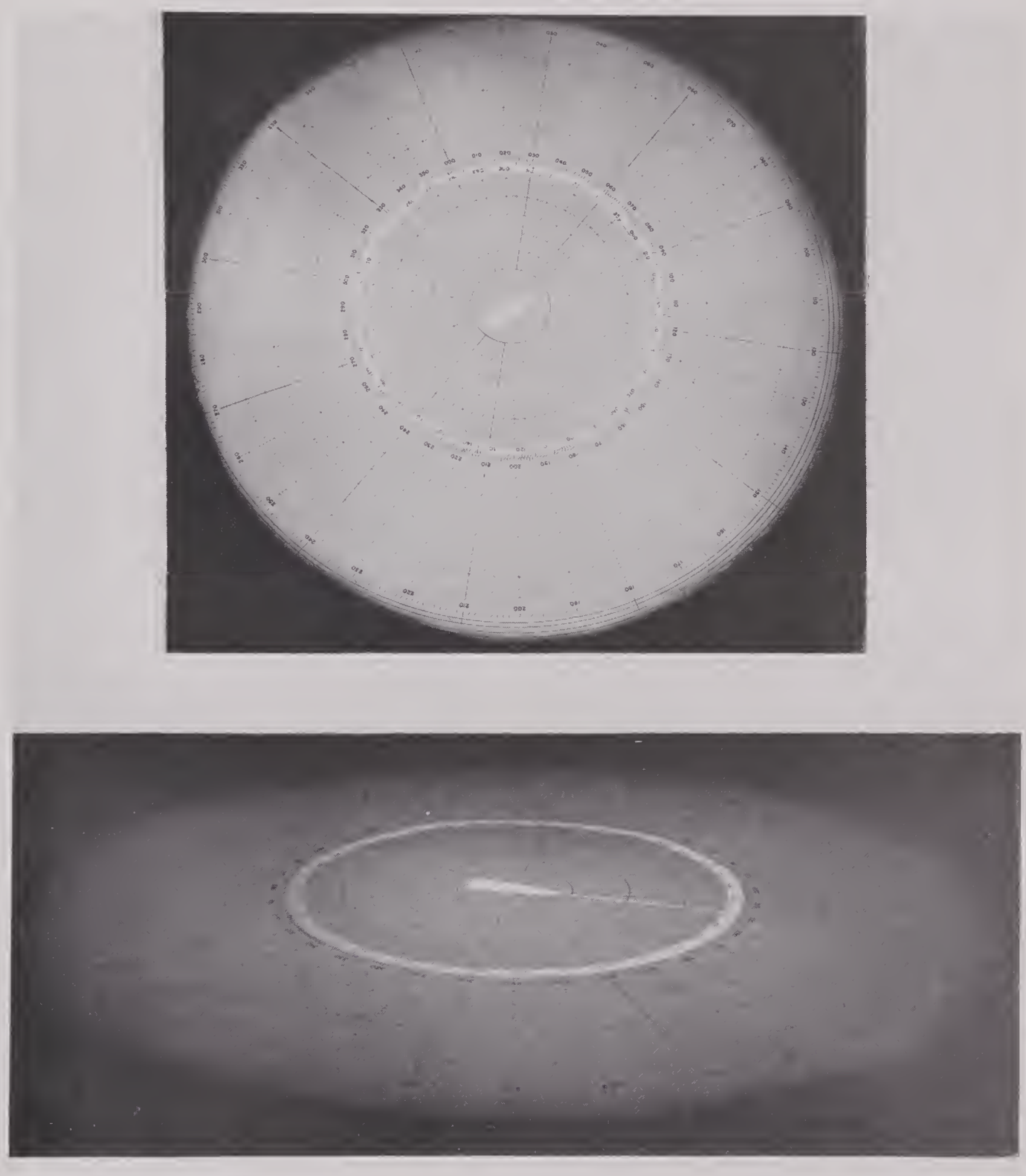


FIGURE 14. Photograph of horizontal plotting table showing own ship's course projection.

Details of the operational and constructional features of the TPG and TPR have been compiled in report form.[1a] The laboratory models of the TPG and TPR were tested and inspected at the Naval Research Laboratory, and many of the important features were incorporated in a new design proposed by the Navy.

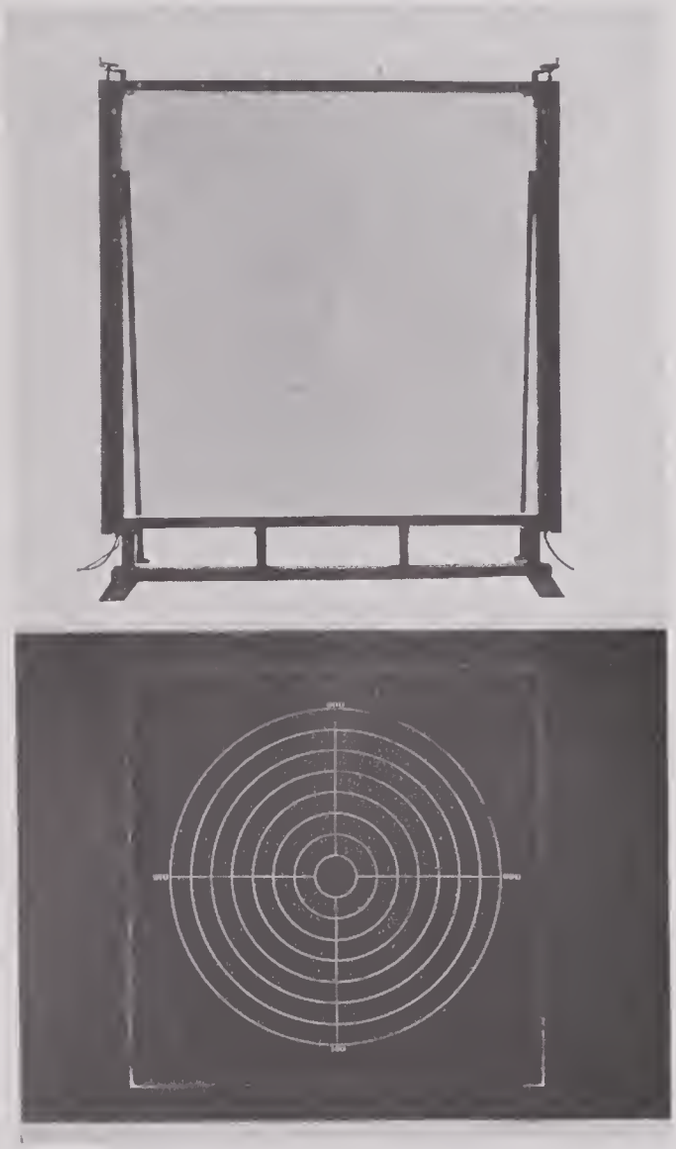


FIGURE 15. Edge-lighted Lucite display board (5x5 ft). The lower photograph was taken in the dark.

19.3 A POINT-SOURCE PROJECTOR FOR USE ON DEAD-RECKONING TRACERS

Facilities for rapid and accurate plotting of radar data on the *dead-reckoning tracer* [DRT] are very important. Universal drafting arms and other auxiliary plotting devices have not proved satisfactory.

In cooperation with Section 634, U. S. Navy Bureau of Ships, a simple lightweight projector was developed at the Electro-Acoustic Laboratory.[2a] The device is designed to be mounted on the pencil carriage of the DRT, and projects a a polar coordinate system 26 in. in diameter onto the plotting surface (see Figure 9). Direct plotting of radar contacts is thereby greatly facilitated. If a plotting scale of 2,000 yd per in. is used, a circle of 13 miles radius about the ship is covered.

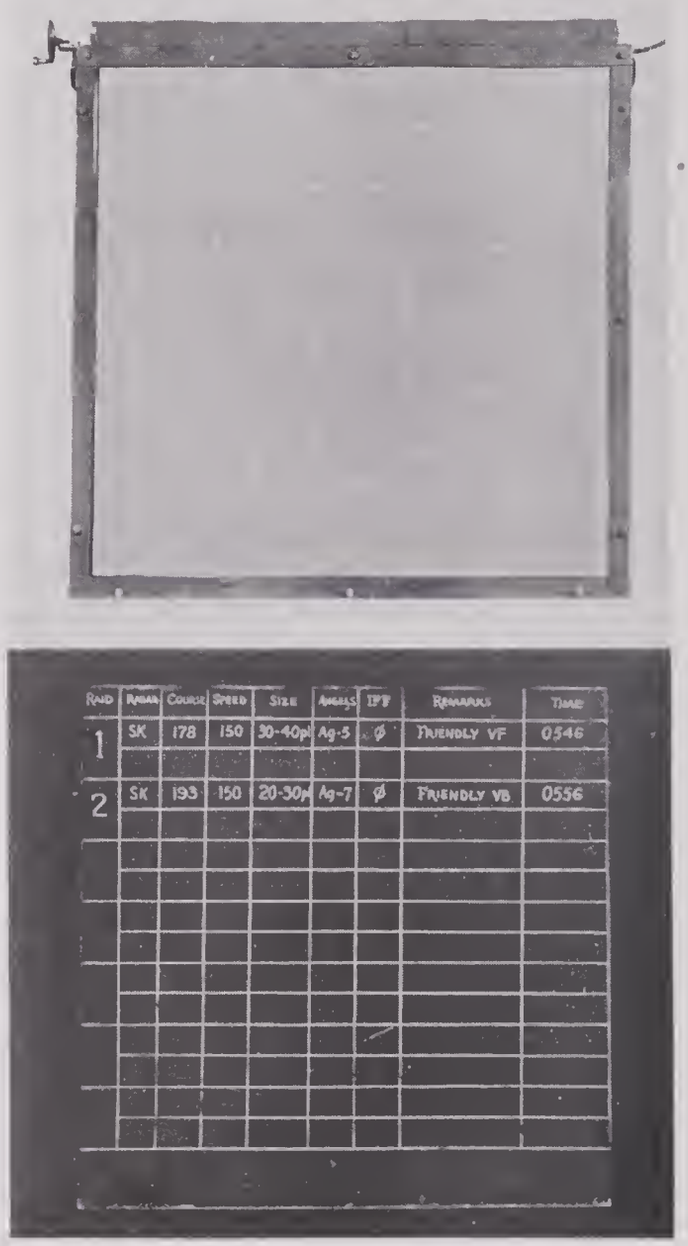


FIGURE 16. Edge-lighted Lucite status board (3x3 ft). The lower photograph was taken in the dark.

To project the image of the coordinate system on the plotting surface a point-source projector is used. It consists of a lightweight case, a glass plate with the coordinate system to be projected, and a tube which supplies the "point" of light of required intensity. Figure 10 shows, diagrammatically, a laboratory model of the

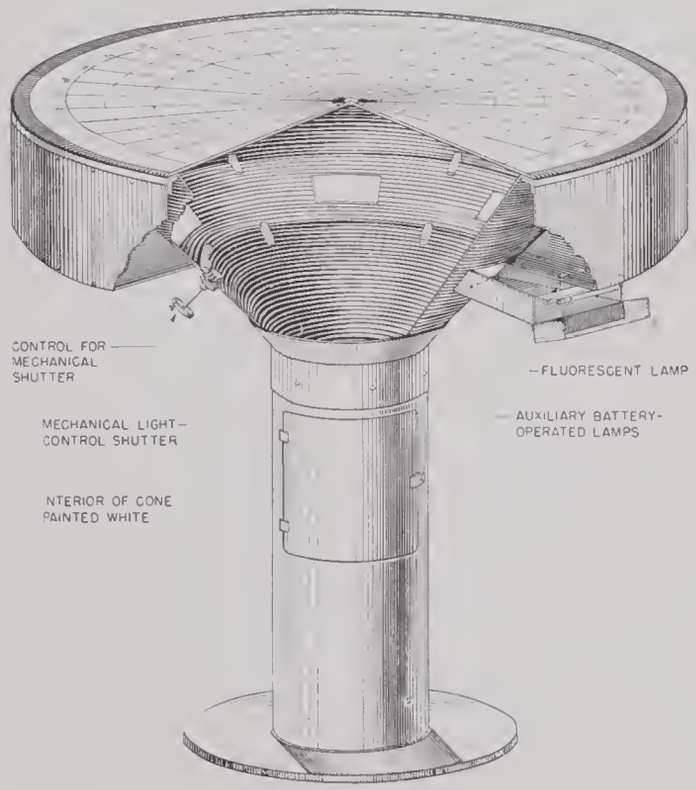


FIGURE 17. Horizontal plotting table with indirect lighting.

projector mounted on the carriage of a standard DRT (Arma). A photograph of the device is shown in Figure 11.

The light source is a gas-filled tube with two electrodes. After a low-impedance ion path between cathode and anode has been started by a surge from the power-supply unit, an arc is maintained between the two electrodes with a voltage drop of about 20 v. The cathode consists of a cylinder of tantalum whose center is filled with zirconium oxide. The area of this core showing on the end face of the tantalum cylinder becomes incandescent under ion bombardment. The anode surrounds the cathode, and by a suitable opening a usable projection angle of 120 degrees is obtained. Figures 12 and 13 show the point-source tube in detail.

In its present form the power consumption of the tube is about ½ w, and its average lifetime is about 500 hours. A suitable power-supply unit with arrangement for starting the arc is discussed in the report mentioned above.[2b] Other applications for a light source of this type suggest themselves. As an example, Figure 14 shows a photograph of the image of a true relative bearing conversion scale projected on a plotting table of conventional design by means of the light source.

19.4 LIGHTING OF PLOTTING AND DISPLAY SURFACES

The lighting problem on shipboard is an important one. Proper lighting of the plotting and display surfaces goes a long way toward efficient operation of information systems as a whole. A high level of light intensity is not permissible, owing to the presence of cathode-ray tubes used as remote radar indicators. It is, therefore, necessary that the plotting and display boards be designed for use in low ambient light intensity.

The device of using a suitable plastic as plotting surface and introducing light through the edges by means of fluorescent tubes has been shown to yield good results for vertical plots.[2] Figures 15 and 16 show vertical plots of the display and status board types photographed in light and in the dark. The lines traced with grease pencil are clearly visible.

In addition to vertical plotting surfaces, consideration must be given to the horizontal plotting tables. It is essential that the illumination of the plotting surface be diffuse for minimum eye strain. A method of indirect lighting has been devised at the Electro-Acoustic Laboratory. Fluorescent lamps are spaced around the table in such a way that their light is introduced into the inside of the metal cone below the plotting surface by suitable openings. No direct light can thus reach the plotting surface. Flat white paint is used to coat the inside of the cone to achieve diffusion. Mechanical shutters control the light intensity.

Figure 17 shows a schematic section through a plotting table using indirect lighting of the type described.

GLOSSARY

α. Absorption coefficient.

ACCOMMODATION, SPEED OF. In vision, the speed with which the shape of the eye lens changes as the individual refocuses for a different distance.

AD. Altitude decrement, the ratio of the response of a device at sea level to its response at altitude. The response may be expressed in terms of the sound pressure of an earphone, the voltage of a microphone, or the gain of an amplifier, or of a whole communication system. In that case, it may include the effects of altitude on listener and talker.

AI. Articulation index, an empirical function with additive properties, directly related to the articulation score obtained with a given crew of talkers and listeners, using specified speech material.

AIRSPEED, TRUE. Speed of aircraft relative to the air at a large distance.

AMPLITUDE DISTORTION. A device is said to introduce amplitude or nonlinear distortion if the output is not linearly related to the input.

ANECHOIC CHAMBER. Room free from acoustic wall reflections (echoes).

ARTICULATION. The articulation test is a procedure by which a quantitative measure of the intelligibility of speech can be obtained. The articulation score is the percentage of the items received correctly by the listeners for the conditions under test. Syllable, word, or sentence articulation scores are obtained with meaningless syllables, isolated words, or complete sentences, respectively, as the test material.

ARTIFICIAL EAR. Cavity with calibrated test microphone simulating the human ear for testing earphones.

ARTIFICIAL VOICE. Sound source simulating a human talker for testing microphones.

ASA. American Standards Association.

ATB/ARB. Naval aircraft transmitter and receiver.

ATTENUATION. The number of decibels by which the sound intensity is reduced by the structure in question. This may be a panel, wall earphone socket, or noise shield.

AUDIOGRAM. Auditory threshold as measured by a standard audiometer, plotted as a function of frequency.

AUTOMATIC ANNUNCIATOR. A device which announces altitude, airspeed, or similar instrument indications directly to the pilot's earphones.

B. Spectrum level of noise at the listener's ear.

B_s. Spectrum level of speech at the listener's ear.

BHP. Brake horsepower—the power delivered by the engine to the propeller, in hp. In multi-engine aircraft the power per engine is always given.

BINAURAL. Stimulation presented to both ears.

BOUNDARY LAYER. Region of the airstream near the surfaces bounding the flow.

BTL. Bell Telephone Laboratories.

CARRIER SENTENCE. In articulation testing, a phrase, such as "you will write——," which the talker uses to monitor his voice level and which the listeners use as a signal for listening.

CENTER CLIPPING. A type of amplitude distortion. A center-clipping or peak-pass circuit will pass only peaks exceeding some fixed value.

CIRCUMAURAL SOCKET. Earphone socket designed to seal against the head around the ear with a cavity large enough to accommodate the pinna.

COUPLER. (1) Artificial ear. (2) Earphone socket or tip.

CRITICAL BANDWIDTH. The width of the band of noise (of the continuous type of spectrum) necessary for the total energy in the band to be equal to that of a given pure tone at its masked threshold. The critical bandwidth is conveniently expressed in decibels, K.

DRT. Dead Reckoning Tracer.

$E = \sqrt{\frac{RT^x}{\sigma_m}}$. Efficiency index of a soundproofing structure.

EARPHONE SOCKET. Socket or cushion which couples acoustically the earphone unit to the ear.

EAR WARDEN. An aural protective device designed to exclude intense and undesirable noises from the ear. Ear plug.

EQUAL-ARTICULATION CONTOUR. Contour relating the bandwidth of a voice communication system to the level of speech, with articulation score as parameter.

EQUIVALENT FREE-FIELD SOUND PRESSURE. The free-field sound pressure of a signal which sounds as loud (to a normal listener seated in a quiet room free from acoustic wall reflections) as a signal of the same type and frequency introduced over earphones. The free-field sound pressure is measured at the listener's location, and is generated by a sound source located some distance in front of him.

f. Frequency, in cycles per second (c).

FLYBAR. Instrument flying by auditory, rather than visual, signals.

FREQUENCY DISTORTION. A device is said to introduce frequency distortion if the efficiency with which different frequencies are transduced is not constant.

GAIN FUNCTION. Function relating the articulation scores to the gain of the system under test, for quiet or fixed conditions of noise.

HANDSET. A combination of a single earphone and microphone joined by a suitable handle.

HCCR. Central Communications Research, Cruft Laboratory, Harvard University.

HEADSET. A combination of two earphones and earphone sockets (cushions or tips) and a suitable headband and cord.

HEAL. Electro-Acoustic Laboratory, Cruft Laboratory, Harvard University.

HEARING LOSS. Shift in the auditory threshold from the normal, or reference value.

HPAL. Psycho-Acoustic Laboratory, Harvard University.

IAS. Indicated airspeed—the reading of the airspeed indicator in mph.

IMPERVIOUS SEPTUM. Layer of material with infinite flow resistance.

IND. ALT. Indicated altitude—the reading of the altimeter without correction, in feet.

INSERT TIP. Earphone socket designed to be inserted into the auditory canal.

INTERPHONE. A voice communication system consisting of microphone, earphones, and intermediate transmitting network.

JASA. Journal of the Acoustical Society of America.

JAMMING. Countermeasures used to deprive the enemy of information normally received via his communication systems.

K. Width of a critical band of frequencies, in decibels.

LISTENING TEST. Test of listening ability, usually administered to a group of listeners.

LOUDNESS. That aspect of auditory sensation in terms of which sounds may be ordered on a scale running from "soft" to "loud." Loudness is chiefly a function of the intensity of a sound, but is also dependent upon the frequency and the composition. The unit is the sone.

LOUDNESS LEVEL. The loudness level, in phons, of a sound is numerically equal to the intensity level in decibels of a pure tone of 1,000 c which is judged by the listeners to be equal in loudness.

M. Masking.

MACH NUMBER. Ratio of stream velocity to sonic velocity at a point in the airstream.

MAF. Minimum audible field. Free-field sound pressure level at threshold in quiet at the observer's location.

MAP. Minimum audible pressure. Threshold sound pressure level in quiet, measured at the observer's eardrum.

$MF = \frac{n_s}{\sigma_s}$. Merit factor of a soundproofing structure.

MASKING. Masking is defined as the number of decibels by which a listener's threshold of audibility is raised in the presence of another sound.

MAXIMUM CRUISING POWER. Power condition of an airplane engine corresponding to about 60 to 70 per cent of rated power.

MEL. The unit of pitch. It is so defined that a 1,000-cycle tone 40 decibels above threshold has a pitch of 1,000 mels. (The mel is a so-called "subjective" unit.)

MILITARY POWER. Power condition of an airplane engine corresponding to the maximum power available. About 110 to 125 per cent of rated power. It cannot be maintained for more than several minutes.

MONAURAL. Stimulation presented to one of the two ears.

n_3. Difference between actual average attenuation N_3 and average attenuation predicted by weight law, in decibels, at 3,000 cycles.

n_5. Difference between actual average attenuation N_5 and average attenuation predicted by weight law, in decibels, at 5,000 cycles.

$N = 10 \log 1/\tau$. Attenuation of an acoustic structure, in decibels.

N-N. Noise-to-noise: Both announcer and listeners are located in an ambient noise field.

N-Q. Noise-to-quiet: The announcer is located in an ambient noise field, while the listeners are in the quiet.

NACA. National Advisory Committee for Aeronautics.

NDRC. National Defense Research Committee.

NEAR POINT OF VISION. The shortest distance at which a subject can accurately focus his two eyes on an object.

NOISE-CANCELING MICROPHONE. A microphone responding primarily to the pressure gradient in a sound wave. It is designed for use close to the talker's lips ("lip" microphone).

NOISE-PEAK LIMITERS. Peak-clipping devices designed to eliminate static peaks from radio receivers.

NOISE SHIELD. An enclosure worn over the talker's lips and sealing well to his face to shield the microphone contained in it from unwanted external sounds.

NORMAL CRUISING. Power condition of an airplane engine well below maximum cruising power.

$\omega = 2\pi f$. Angular frequency in radians per second.

ORTHOTELEPHONIC GAIN [OG]. The number of decibels by which the signal at the listener's ear, when the communication system in question is being used, differs from the signal which would exist at his ear if the talker, talking in the same way, were communicating with him over a free path of air one meter in length in a quiet room free from acoustic wall reflections.

ORTHOTELEPHONIC REFERENCE SYSTEM. A communication system, consisting of a normal listener and talker who face each other at a distance of one meter in a quiet room free from acoustic wall reflections.

OSRD. Office of Scientific Research and Development.

PAPER FORM BOARD. (Revised Minnesota Form Board Test.) The testee is required to manipulate geometric shapes to fill a prescribed area. The score is the number of correct answers given in a limited period of time.

PEAK CLIPPING. A type of amplitude distortion. A peak-clipping or peak-limiting circuit does not pass peaks exceeding some fixed value.

PEAK FACTOR. Ratio of peak to rms value in a complex wave.

PHON. A unit for measuring the loudness level of a tone.

PHONEME. Minimum distinctive unit of speech sound.

PHONETIC ALPHABET. A set of symbols representing speech sounds. In military usage, the term refers to a list of words used in speech communication to identify the individual letters of the alphabet (*A*ble, *B*aker, *C*harlie, etc.)

PITCH. That aspect of auditory sensation in terms of which sounds may be ordered on a scale running from "low" to "high." Pitch is chiefly a function of the frequency of a sound, but is also dependent upon the intensity and the composition. The unit is the mel.

PREMODULATION CLIPPING. Clipping the peaks of the speech signal prior to modulation of the amplitude of an r-f carrier wave.

PRESSURE ALTITUDE. Altitude in feet, defined by the U. S. Standard Atmosphere, at which the atmospheric pressure is equal to the given pressure.

PRF. Pulse repetition frequency.

PROPELLER-TIP PASSAGE FREQUENCY. Frequency at which the propeller tips pass a fixed plane through the propeller axis, in cycles per second.

PSYCHOMOTOR EFFICIENCY. The efficiency, measured by speed or by the number of errors, of the muscular responses to cerebral processes.

PURSUIT METER. An instrument for measuring eye-hand coordination, which combines (1) a stimulus target, the movements of which the subject endeavors to follow, and (2) an integrating meter which registers the amount of coordination error in terms of amplitude or time or both.

Q-N. Quiet-to-noise: the announcer is located in the quiet, while the listeners are in an ambient noise field.

Q-Q. Quiet-to-quiet: both announcer and listeners are located in the quiet.

R. Flow resistance.

$R = \lim_{v \to 0} p/v$. Flow resistance, in grams per square centimeter per second.

RATED POWER. Power condition of an airplane engine corresponding to maximum *continuous* power available. Also called maximum (continuous) power.

REAL-EAR RESPONSE. The real-ear response of an earphone is the ratio of the equivalent free-field sound pressure to the voltage applied across the terminals. The determination of the equivalent free-field pressure requires a judgment of equal loudness for tones of various frequencies.

REAL-VOICE RESPONSE. The real-voice response of a microphone is the ratio of the microphone output voltage to the free-field sound pressure measured one meter in front of the lips of a normal talker wearing the microphone in the normal manner. Usually the talker uses a representative speech sample and the real-voice response is determined in contiguous bands of frequencies by suitable analyzing equipment.

RELIABILITY. Reliability is indicated by the degree of constancy obtaining between repetitions of the same measurements.

REVERBERATION CHAMBER. Chamber with hard, sound-reflecting walls.

RMS. Root-mean-square.

RPM. Revolutions of the engine crankshaft per minute.

RRU. Radio repeat unit.

σ. Surface density of total structure.

σ_s. Surface density of total structure, less Dural skin.

σ_m. Surface density of absorbing material alone.

SCHLIERE. Striation, due to local changes of index of refraction, made visible by schlieren (striation) method.

SCHLIEREN (STRIATION) METHOD. Optical method to allow visual observation of schlieren (striation).

SEMI-INSERT TIP. Earphone socket designed to seal against the opening of the auditory canal.

SENSATION LEVEL. The number of decibels a sound is above the threshold of audibility.

SENSITIVITY OF THE EAR. See *threshold*.

SIMULTANEOUS RANGE. A type of radio range transmission which provides both range and voice signals on a single channel. Also called 5-tower range or TL range.

SL. Spectrum level. The rms level of a sound of the continuous spectrum type, in bands one cycle wide. It is expressed in decibels relative to the standard reference level.

SONE. A unit of loudness. It is defined as the loudness of a 1,000-cycle tone 40 decibels above the normal threshold. (The sone is a so-called "subjective" unit.)

SONIC VELOCITY. Velocity of sound.

SOUND-POWERED TELEPHONE. A telephone system relying for operation only on the energy supplied by the talker's voice.

SPAN OF APPREHENSION. The amount or complexity of perceptual material apprehended during a single brief presentation.

SPECIFIC FLOW RESISTANCE. Flow resistance per unit thickness of sample.

SPECTRUM. The spectrum of a sound of the continuous spectrum type is a plot of its spectrum level as a function of frequency. The spectrum of a wave containing a finite number of discrete components consists of a plot of the level of these components versus frequency. By extension, the spectrum of a wave containing both discrete and continuous frequency components is a combination of both types of spectra.

SPI. Sonic Positioning Indicator.

SPL. Sound pressure level. The sound pressure, in decibels, relative to the standard reference level of 0.0002 dyne per square centimeter.

SR. Sonic Receiver.

STASI. Sonic True Airspeed Indicator.

STATIC CANCELING CIRCUITS. Noise-reducing devices in radio receivers which operate on the principle that static (impluse noise) in adjacent channels can be used (by proper phase adjustment) to cancel out the impulse noise in the pass band of the receiver.

STIMULATION DEAFNESS. A loss in hearing, usually temporary, resulting from exposure to intense sounds.

STREAM VELOCITY. Velocity at a point in the airstream.

SUBSONIC VELOCITY. Velocity smaller than the sonic velocity.

SUPERSONIC VELOCITY. Velocity greater than the sonic velocity.

SUPRA-AURAL SOCKET. Earphone Socket designed to seal flat against the pinna.

τ. Transmission coefficient.

T. Thickness of sound-absorbing material.

THRESHOLD. The *threshold of audibility* at a given frequency is the minimal value of sound pressure which produces an auditory sensation. (It can be expressed in terms of MAP or MAF.) The *masked threshold* is the minimal value of sound pressure necessary to hear the masked sound in the presence of a masking

sound. The *differential threshold* (for frequency or intensity) is the minimal increment in the stimulus necessary to produce a just noticeable difference in sensation. *Speech thresholds* are obtained by adjusting the speech level (1) until the presence of speech is just noticeable although not intelligible (*threshold of detectability*), (2) until the gist of the speech is understood, although individual sounds or words are not all heard (*threshold of perceptibility*), or (3) until every sentence and phrase is understood without perceptible effort (*threshold of intelligibility*).

THROAT MICROPHONE. A microphone operated directly by the vibrations of the larynx during speech.

TILT. A type of frequency distortion. A circuit having a "tilted" frequency response that rises or falls at a constant rate over the entire range of frequencies under consideration.

TINNITUS. An auditory disorder characterized by a ringing, buzzing, or rushing sound in the ear in the absence of any external stimulus. Occurs frequently following exposure to loud sounds.

TOLERANCE THRESHOLDS. The maximum sound pressure level which a listener can tolerate. *Thresholds* of *discomfort*, of *tickle*, and of *pain* can be distinguished.

TPG. Timing Pulse Generator.

TPR. Timing Pulse Receiver.

U. S. STANDARD ATMOSPHERE. Typical state of the atmosphere found in the continental U. S., defined in Publication No. 538, National Advisory Committee on Aeronautics.

VALIDITY. The extent to which a test or other measuring device measures what it purports to measure.

VOICE TUBE. A tube forming an acoustic link between the mouth of a talker and the ear of a listener.

Z. The effective level of a masking noise. *Z* is the difference in decibels between the pure-tone threshold in quiet and the total energy level in the critical band of noise frequencies.

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Chapter 2

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OSRD APPOINTEES

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Chiefs

George R. Harrison
Paul E. Klopsteg

Deputy Chiefs

E. A. Eckhardt
Paul E. Klopsteg

Technical Aides

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Francis L. Yost

Clark Goodman
John A. Hornbeck (WOC)

Members

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Theodore Dunham, Jr.
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Harvey Fletcher

William E. Forsythe
George R. Harrison
Herbert E. Ives
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Clifford Morgan

Members

Hallowell Davis
Floyd A. Firestone

Vern O. Knudsen
S. Smith Stevens

E. C. Wente

Contract Number	*Name and Address of Contractor*	*Subject*
NDCrc-52	National Academy of Sciences Washington, D. C.	Studies and investigations in connection with noise levels, the effects of noise and methods and materials for noise reduction in combat vehicles.
NDCrc-79	Harvard University Cambridge, Massachusetts	Studies and experimental investigations in connection with physiological and psychological effects of and the development of protection against sounds.
NDCrc-110	The Western Electric Company, Incorporated New York, New York	Studies and experimental investigations in connection with the development of sound sources and arrangement of such sources for producing high directivity of the sound beam radiated.
NDCrc-128	The Regents of the University of California Berkeley, California	Studies and experimental investigations in connection with the development of sound sources.
OEMsr-38	National Academy of Sciences	Studies and experimental investigations in connection with the feasibility of developing a directional acoustic microphone for plane following.
OEMsr-39	National Academy of Sciences	Studies and experimental investigations of noise levels, effects of noise, methods and materials for noise reduction in combat vehicles, and such other aspects of air space or water borne sound including problems of underwater submarine detection as may be requested in writing by the Contracting Officer prior to July 1, 1942.
OEMsr-119	Harvard University Cambridge, Massachusetts	Studies and experimental investigations in connection with physiological and psychological effects of and the development of protection against sounds.
OEMsr-146	Western Electric Company, Incorporated New York, New York	Studies and experimental investigations in connection with (i) the development of sound sources and arrangement of such sources for producing high directivity of the sound beam radiated, including development of a siren type of sound projector and the testing of the projector developed, (ii) the design and development of an air raid warning siren, smaller than the high directive siren but employing the principles thereof, and (iii) the delivery and testing, in various parts of the United States as the Contracting Officer or his authorized representative shall direct, of ten (10) models of such sirens as may be developed hereunder.

Contract Number	*Name and Address of Contractor*	*Subject*
OEMsr-172	American Locomotive Company Auburn, New York	To supply one (1) centrifugal air compressor, consisting of a ninety-five (95) horsepower Ford engine, complete with starter, fuel tank, fan, radiator, etc., a gear box and gears, and a blower with air filter, said compressor to be capable of delivering two thousand five hundred (2,500) cubic feet per minute at a pressure of five (5) pounds.
OEMsr-176	Western Electric Company, Incorporated New York, New York	Studies and experimental investigations in connection with the development of methods and apparatus for reducing the static coming from the radio set and the noise coming from the engines of the plane during the reception of speech by airplane pilots.
OEMsr-197	The Trustees of the Stevens Institute of Technology Hoboken, New Jersey	Studies and experimental investigations in connection with the determination of the effect of noises on man and the development of means and methods for producing such noises.
OEMsr-658	President and Fellows of Harvard College Cambridge, Massachusetts	Instrumental and psychological studies and experimental investigations in connection with (i) sound control in combat vehicles and adequacy and efficiency of interphone equipment, (ii) acoustical investigations involving similar equipments and techniques, (iii) acoustical methods of position determination of gliders towed by planes, (iv) tests of samples and new models of microphones, head phones and interphone equipment, (v) development of tests and methods for selecting and training voice communications personnel, (vi) efficiency and adequacy of standard and experimental hearing aids for service incurred deafness among military personnel, (vii) the development of a sonic method for the indication of the true airspeed of aircraft, and (viii) interior shipboard communications and Combat Communication Centers and the development of devices employed therein.
OEMsr-849	Chrysler Corporation Detroit, Michigan	Studies and experimental investigations in connection with the development of a siren-type intense sound source suitable for airplane mounting.

Contract Number	*Name and Address of Contractor*	*Subject*
OEMsr-908	Western Electric Company, Incorporated New York, New York	Conduct studies and experimental investigations in connection with the (i) development of a loud-speaking system capable of being placed manually and suitable for use at sea or on land for purposes of deception and decoy and for conveying intelligence, (ii) development, for further manufacture, of ten improved and lighter weight loud-speaking systems for use in military operations (junior heaters), and (iii) development of three loud-speaking systems suitable for installation in electrically driven torpedoes.
OEMsr-1178	Mine Safety Appliances Company Pittsburgh, Pennsylvania	Studies and experimental investigations in connection with the production of (i) one four-cavity mould for small tips, (ii) two four-cavity moulds for medium tips, (iii) one four-cavity mould for large tips, and (iv) 500 Harvintips of assorted sizes.
OEMsr-1201	Central Institute for the Deaf, a Missouri Corporation St. Louis, Missouri	Studies and experimental investigations in connection with (i) the development of instruments and methods for aiding impaired hearing incurred in military service, and (ii) the development of tests for the evaluation of such instruments and methods when used with partially deafened individuals.
OEMsr-1210	Pennsylvania State College State College, Pennsylvania	Studies and experimental investigations in connection with (i) the development and testing of equipment and methods for utilizing high frequency sound waves for communication in warfare, and (ii) the supersonic acoustics of jungle terrains.
OEMsr-1230	General Electric Company Schenectady, New York	Studies and experimental investigations in connection with the development of three electrically driven torpedoes containing suitable loudspeaking systems which will perform specified functions at the end of a prescribed run of the torpedo.
OEMsr-1240	President and Fellows of Harvard College Cambridge, Massachusetts	Instrumental and experimental investigations in connection with (i) interior communication facilities, instruments and recording devices for use on ships of the Navy, (ii) surveys and studies of efficient means for displaying radar and other information to ship's personnel, (iii) methods and materials for acoustically quieting and air conditioning operations rooms, and (iv) the construction of experimental equipment to demonstrate and test the improvements enumerated herein.

Contract Number	*Name and Address of Contractor*	*Subject*
OEMsr-1300	National Organ Supply Company Erie, Pennsylvania	Studies and experimental investigations in connection with (i) the production of 1250 pairs of small, 2500 pairs of medium, and 1250 pairs of large V-51R ear wardens, together with the required number of applicators and cases, and the moulds required for their production, and (ii) the use of thermo-plastic materials for ear defenders and other aural devices used in communication equipment.
OEMsr-1335	The Trustees of Rutgers College in New Jersey New Brunswick, New Jersey	Studies and experimental investigations in connection with the acoustics of jungle terrains.

SERVICE PROJECT NUMBERS

The projects listed below were transmitted to the Office of the Executive Secretary, OSRD, from the War or Navy Department through either the War Department Liaison Officer for NDRC or the Office of Research and Inventions (formerly the Coordinator of Research and Development), Navy Department.

Service Project Number	*Title*
	Army Projects
AC-9	Soundproof Material.
AC-43	Acoustic Device for Determining the Position of a Glider with Respect to a Towing Airplane.
AN-10	Research in Problems of Rehabilitation of the Deaf.
CE-36.02	Reduction of Sound and Shock in Fortification Structures.
CE-39	Weapons Simulators.
CWS-18	Determination of the Effect of Noise on Men and Development of Means and Methods of Producing Such Noises.
CWS-28	Study of Acoustical Properties of Gas Masks.
CWS-28 Ext.	Films of Better Acoustic Properties for Diaphragm Masks.
OD-110	High-Intensity Expendable Sound Source.
SC-4*	Sound Research.
SC-47	Deceptive Sounds.
SC-48	Microphone and Headset Studies.
SC-105	Acoustics of the Jungle.
SC-118	Psychological Factors in Communication Jamming.

* This work is not discussed in Division 17 STR. For details the reader is referred to the NDRC Bi-Monthly Summaries. The project is listed here for completeness of project information.

	Navy Projects
N-109[a]	Assistance in Selection and Training of Personnel Using Voice Communication Systems.
N-118 Ext.	Assistance in Program of Classification and Training of Telephone Talkers at Submarine Base, New London.
NA-108	Speech Intelligibility of Aircraft Intercommunication Equipment.
NA-108 Ext.	Assistance in the Preparation of an Operational Use Manual.
NA-108 Ext.	Determination of the Optimum Time Constant of the Audio Limiter Circuit of the AN/ARC-2 Equipment.
NA-118	Noise-Making Device for Naval Aircraft.
NA-170	Sonic True Airspeed Indicator.
NA-183[b]	Sonic Control of Guided Missiles.
NR-104	Heater.
NR-107	Development of Supersonic Whistle and Receiver.
NS-191	Flat-Type Headset Receiver.
NS-262	Sound Transmission in Air.
NS-315	Noise Reduction in LVTs.
NS-315 Ext.	Construction of Two Experimental Audio Amplifiers for use in LVTs.
NS-341	Countermeasures to Enemy Detection of Radar.
NS-343	Interior Communications and Associated Equipment.
NS-343 Ext.	Study of the Conversion of a VG Type Radar Repeater.
NS-365[c]	Investigations of Principles Underlying the Maximizing Communication Intelligibility.
SAC-52	Quieting of Outboard Motors.

[a] Division 17 cooperated in work on this project, but N-109 was not formally assigned to it.

[b] Division 17 was associated with this project in an advisory capacity only. The project was not formally assigned to Division 17.

[c] This project was assigned to Divisions 13 and 17 in cooperation.

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